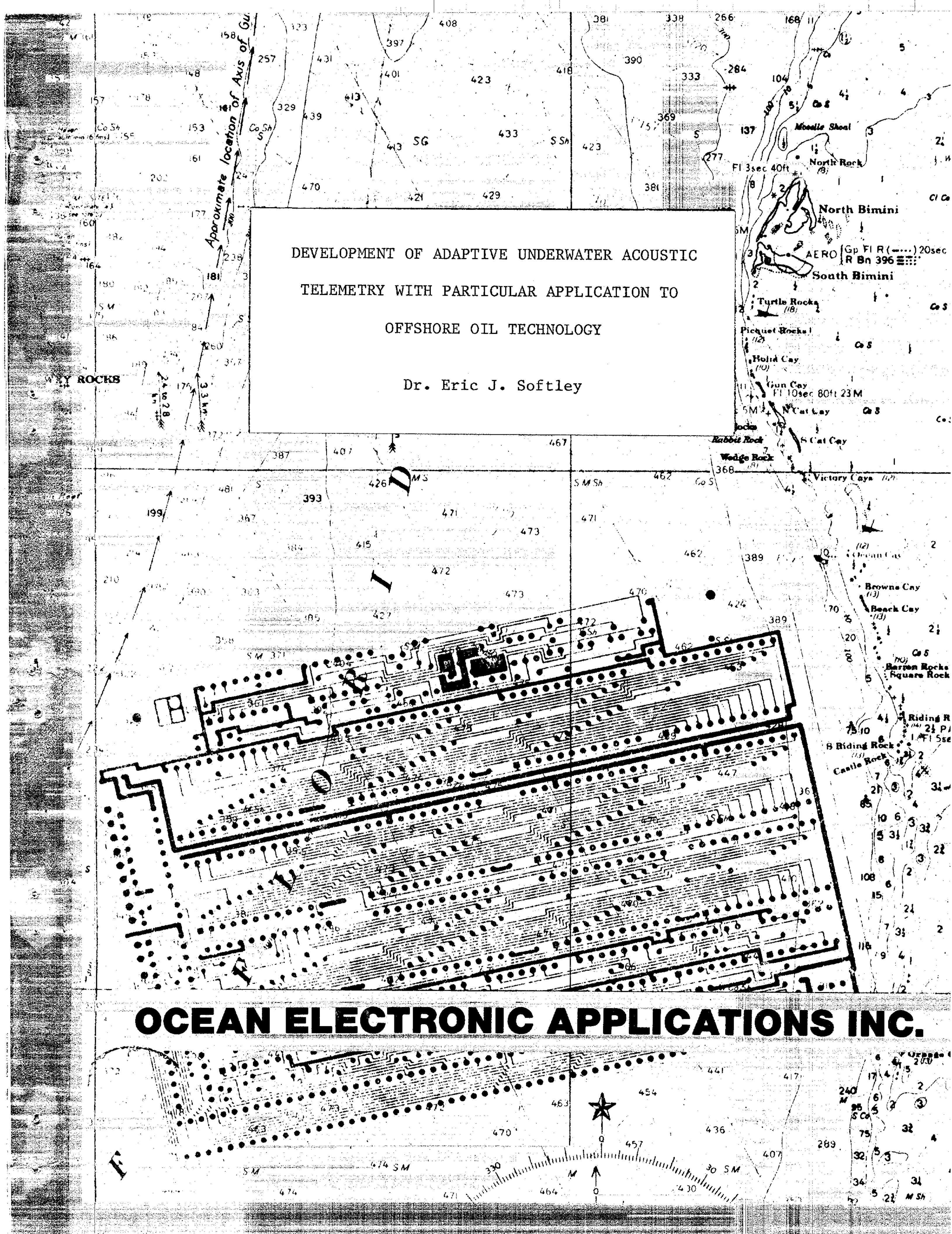

DEVELOPMENT OF ADAPTIVE UNDERWATER ACOUSTIC TELEMETRY WITH PARTICULAR APPLICATION TO OFFSHORE OIL TECHNOLOGY

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Acknowledgements.

The author wishes to acknowledge the diligent efforts of Mr.Andy Lyn in the development of the concept described in this report. He also wishes to thank Miss Kim Rowell and Mr Jens Heilemann for all their efforts in creating the hardware and performing the experimental studies. Finally the creative support and encouragement of Mr. John Gregory of the Minerals Management Service was a crucial ingredient without which nothing would have been accomplished.

1. INTRODUCTION

The need for data telecommunications has become increasingly apparent as the computer becomes a more frequent component in everyday use. This is true in many instances including the offshore oil industry. The low powered microcomputer with a multiplicity of capabilities is obviously useful for many data gathering and system control functions. Well head measurements, well head valve controls, offshore platform monitoring, command and control of remote vehicles and data telemetry such as compressed video information from these vehicles are all examples where the data link is a critical part.

For many data communication tasks above water hardwired systems or, where this is impractical, RF telemetry are often used. In underwater applications neither of these approaches is attractive and acoustic telemetry is a viable alternative (references 1 - 4). Hence there is a growing interest in the use of acoustic data telemetry in the applications mentioned above.

Some of these applications can have rather simple acoustics and have been tackled quite effectively by straightforward systems. The vertical or near vertical path is one such example. In many instances, however, the acoustics are complicated by the presence of multiple paths with signals of camparable strength (known simply as the multipath problem). Also the acoustic paths may fluctuate with time producing signal variation and jitter of transmitted data.

Additionally the ambient noise can be much more significant in locations of practical interest. Drilling noise, pump noise, ship noise (especially propeller noise) and noise from other engineering activities can be very high and make acoustic communictions more difficult.

The impact of many of these problems can be adjusted by changing the acoustic parameters of the system. The fundamental physical phenomenon is the increased attenuation of the signal with increasing frequency. This decreases the amplitude of any signal but also decreases the amplitude of the multipath signal. In instances where the difference between the two also increases separation of the two could be influenced by increasing the frequency.

A second factor is that signals bouncing from the ocean floor experience increased attenuation as the frequency increases. Again the frequency is a useful variable in the system design.

Note that if the system itself had a high tolerance to multipath level (small signal difference) then the parameter adjustments can be more effectively used. As observed increasing the frequency improves the multipath separation but degrades the ambient noise tolerance. The best choice of frequency will be a balance of the two and with more multipath tolerance a higher frequency can be used.

A third basic factor is that as the acoustic frequency is increased then the data rate can be increased. Hence for given input power the telemetry becomes more efficient at higher frequency. This factor is strongly influenced by environmentally induced bit jitter however. This bit jitter appears to originate from two sources. Acoustic paths which involve reflection from the ocean surface or which pass through areas of thermal turbulence incur path length changes and hence the bit jitter.

It has been the authors experience thet the different decoding methods all represent a compromise in combating the various problems. For example a method of decoding which is multipath tolerant will not usually allow high data rates. Similarly a linear high data rate decoder will not tolerate high multipath levels. A decoder which can decode data with high ambient noise levels is likewise less tolerant to multipath and more limited in data rate.

The last factor is that if high signal to noise occurs and the other factors are adequate then input power could be reduced to improve the efficiency of the system.

Hence the basic concept of the adaptive acoustic telemetry system presented here is to design a system which can decode signals with small signal to multipath margins and, using that system, to vary the parameters

1. Acoustic frequency.
2. Data rate.
3. Power level.
4. Data decoder type.

to provide the highest data rate at the highest efficiency and with minimum error. Since the acoustic situation is not known, a priori, then the system must also analyze the acoustic link and maintain a dialogue between the transceiver elements in order to make the parameter adjustments.

The system that has been developed under this contract has the goals outlined above. The usefulness of the differing parameters and the applicability to the offshore oil industry is examined by analyzing several scenarios. The acoustic signal strength is examined by calculating ray paths for the scenarios and, where possible, using engineering data on the noise levels.

The scenarios chosen for analysis are:

1. Interrogation of ocean floor located instrumentation from the surface.

2. Interrogation of platform instrumentation (on the underwater structure) from the deck.

3. Communication with free swimming inspection vehicles.

4. Interrogation of ocean floor mounted instrumentation (pipeline) from an offshore platform.

From these analyses it is intended that certain procedures be identified for the adjustment of the acoustic parameters. These procedures will provide a starting point for the adaptability software development.

2. BASIC CONCEPTS OF ADAPTIVE ACOUSTIC TELEMETRY

The rationale for parameter variation has been discussed in the introduction. The parameters to be considered as variable are acoustic frequency, data rate, transmitter power and type of decoder.

The type of modulation and coding will be fixed. From our experiences with variable propagation and high noise, frequency modulation was selected. With digital coding this means that the acoustic frequency (or carrier) is switched between two levels. The switching is not instaneous and it is felt that FM rather than FSK is a better description of the process. The ratio betweem the two frequencies is kept constant. This allows the use of variable center frequency but constant ratio filters in the decoder. The total inclusive bandwidth is not narrow, however the bandwidth in use will be kept reasonably narrow.

While one standard method for improving selectivity of signals over ambient noise is to use narrow bandwidths this is extremely difficult in the ocean. Movement of the transmitting and receiving transducers is often unavoidable and the result is a doppler shifting of the frequency. This could be overcome by tracking of the incoming signal. However multipath signals were present this tracking was not successful in the experiments by the author.

With the fundamental selection of pairs of acoustic frequencies of specified ratio then either a continuously varying system or a set of discrete channels can be employed. Since the acoustic frequency will be unknown to the receiver the second approach is used. This allows a combination of switched capacitor analogue filters together with selectable transverse filters in the receiver design and makes the acoustic frequency selection more attainable.

The actual channel frequencies can vary considerably. In three of the systems created to date four channels with an overall frequency ratio of 4:1 have been used. Eight channels have been used in one system but the time for channel selection can get quite lengthy. For a short range situation as for example in the platform monitoring the frequencies ranged from 60 to 120KHz. For a long range situation as in scenario 4 the frequencies varied from 8 to 27KHz. The selection of frequencies can be roughly derived by examining the attenuation of a typical signal with absorption and spherical spreading. This gives the results shown in figure 1. A typical signal source of 150 dB was used with a system noise of 50 dB to scale the data. Other factors will add to the selection criteria for a real system.

The method of coding is selected from a number of factors. First the coding must be self clocking. This is necessary since the data rate is not known a priori. Secondly the coding must be decipherable with considerable bit jitter. This is introduced from environmental fluctuations as mentioned earlier. There are several candidate coding schemes with the three strongest being split phase (or Manchester), pulse position and pulse modulation. The Manchester coding is the most common since it uses the least bandwidth for a given data rate. It uses transitions as the basic message and since this implies differentation of the signal high signal to noise is desirable. Moreover the clock reconstruction involves both transition polarities and inverted data must be recognized and inverted. With the data rate unknown this is an extra problem which can slow up the data recovery.

Pulse width modulation uses 50% more bandwidth than Manchester but has several advantages for the problems here. The clock is reconstructed from positive transitions alone and the coherence of the bits i.e. the consistency of the duration of successive bits is an important determinant in recognizing incoming data. In addition in high noise situations integral methods can be used for data decoding.

The power level used for transmitting is a simple variable. The design objective is to reach a transmitting level of 170 dB re 1 uPa at 1 meter from the transmitting transducer. If a favorable acoustic situation occurs the signal can be reduced. In practice 3 steps of 10 dB below the maximum are used.

The method of decoding depends on the situation. The primary design objective was to decode data with high multipath signal present. A process involving splitting the incoming signal into two narrow bands corresponding to the upper and lower frequencies is the primary design element. The presence of the signal in each band was determined by using a floating comparator. Since multipath signal become distortions to the presence of each signal the presence detector must be sensitive to the multipath and adjust accordingly. In practice this method works up to a 4 dB signal to multipath ratio. Higher levels of multipath signals cause rapidly increasing data errors.

The primary objection to this method is that since the filters represent high Q circuitry they have a poor transient response and a maximum data rate. Figure 2 shows this limit. For many cases the data rates are still quite useful. For video telemetry a higher data rate is more useful however.

If a high signal to noise and low multipath levels are realized then a simple hard limiter could be used to decode the data. Much higher data rates are achievable, typically better than 10% of the acoustic frequency. Bit jitter problems can still occur however and this method has only limited usefulness.

One method of data decoding which is particularly applicable to PWM coding is useful in cases where high noise levels occur but very little multipath signals exist. It is also more tolerant to bit jitter. It is practical for many poor signal situations. The method uses counters to assess the probability of whether the data coding is high or low. More details on this will be given later. In practical situations error free data has been achieved with signal/noise of -4 dB. Error free here means 1 in 10^6.

The probability decoder does have some data rate limitations but not quite as severe as the transverse filter array decoder. It must be used in conjunction with a linear demodulator.

Figure 2 shows typical data limitations found in the laboratory. In practice better response might be achieved by optimization of the system. However it does show the relative response capabilities. The following table gives some relative merits of the decoders:

Type Decoder	Limiting values for S/N, S/M, Jitter and Data Rate			
	Minimum S/N	Minimum S/M	Maximum Jitter	Maximum Data Rate
Transverse Filter Array	4 dB	4 dB	25%	4% F 3000 Baud.
Hard Limiter	10 dB	20 dB		10% F
Probability Decoder	-4 dB	10 dB	15%	8% F 10000 Baud.

where F is the acoustic frequency. It should be noted that the probability decoder was not attempted at rates higher than 10K baud. 25% jitter means that bit durations can vary $\pm 12.5\%$ during the data transmission.

Hence the primary decoder choice is the transverse filter array with the use of a hard limiter for signals of high signal to noise and signal to multipath and the probability decoder available for poor signal to noise situations.

A basic concept of adaptability is that the transceivers can analyze the acoustic situation. To enable this a process of intercommunications is used. Initially a transmission takes place from one or either transceivers and the second unit uses the transmission for analysis. The acoustic frequency is recognized by scanning and signal and noise analysis The data rate is recognized by using coherence of the bits and evaluation of the preamble. The multipath level is recognized by measurements within the transverse filter array. The receiving unit can then adjust the parameters and telemeter back a reply.

One factor which requires additional information is the noise at the other transceiver. Information on this noise level must be transmitted as part of each message. The noise information can be included in the first transmission but a more useful value is the signal to noise ratio and this implies at least two or three transmissions between the units.

In summary then the adaptability process requires:

Adjustable parameters.
Ability to recognise imcoming message.
Analysis of the acoustics.
Intelligent readjustments of parameters.
Conversation between transceivers.

3. ANALYSIS OF SEVERAL SCENARIOS.

The four scenarios selected for analysis are:

1. Interrogation of ocean floor located instrumentation from the surface.

2. Monitoring of underwater instrumentation on offshore platforms from the deck.

3. Communication with free swimming vehicles near offshore platform.

4. Interrogation of ocean floor mounted instrumentation located in the vicinity of an offshore platform.

Each of these has different acoustic situations and represents differing problems. The approach used in the analysis is to use a temperature profile for a known platform and to use ambient noise measurements for that platform. Acoustic ray traces will allow the calculation of path length, absorption, spreading loss and where applicable bottom reflection losses for each case. The object is to examine each to determine the criteria that an adaptive system could use if it had the opportunity to perform such an analysis.

It is necessary to be quite specific in terms of parameters and assumptions in order to make these analyses. This is in general not a problem because the intent is to identify criteria for adaptability rather than to select operating parameters.

3.1 Scenario 1 - Ocean floor to surface.

Figure 3 shows the acoustic situation. A transceiver is located near the ocean floor. The second transceiver is located on the surface such that the acoustic paths are nearly vertical. The transducer for the second transceiver is suspended from the surface. Clearly acoustic rays reflecting from the surface will add (or subtract) significantly to the direct signal. Therefore in any practical situation the transducer is assumed to have a strong directionality concentrated in the downward direction. In experiments described later a front to back difference of 30 dB was measured and this is assumed in the analysis.

Figure 4 shows results of the calculations for several depths. The R ray is the direct path assuming spherical spreading and absorption losses. The S ray represents the surface reflection from the backward ray. The first multipath is 3B and is a signal which reflects from the bottom and again from the surface, traversing the path three times. The bottom loss will be very dependent on material and assumed to be that given in Urick (ref 5). The noise is also taken from the same reference and integrated over the analogue filter bandwidth and this, in turn, is assumed proportional to frequency.

As expected the multipath signals are not significant and the system must choose a frequency dependent primarily on signal to noise and on decoder type. A linear decoder and hard limiter will work quite well with the frequency dependent on depth and ambient noise. Figure 5 shows the frequency and data rate for the hard limiter decoder and for the probability decoder.

The basic adaptive criteria is:

1. Determine absence of significant multipath

2. Select decoder.

3. Increase frequency until S/N is 20 dB (hard limiter) or 6 dB (probability) detectors.

4. Determine data rate independently from jitter measurements.

Note that 10 dB margin is allowed to cover transient noise possibilities.

The two decoders allow sharply different acoustic frequencies because of the different noise margin requirements. The data rates are slightly higher for the probability decoder as a result. In practice these data rates were never achieved in this program because of the presence of environmentally induced bit jitter.

In some practical situations similar to the above idealized scenario the noise was significantly different. The presence of a passing ship increased the noise, especially below 30 KHz. This would only influence the process above if the the noise was sufficient to create two solutions to the adjustment process above. If the system were in water depths of less than 1000M the system should approach each operating point from a frequency higher than 30 KHz. In deeper waters there is no change to the process and the frequency should be approached from well below 30 KHz. Thus a significant difference in approach will be used depending on the ocean depth.

If the surface unit initiates the adaptive process and is manually controlled the observation of the increased noise or of the noise source is an input to the selection of the initial system parameters. To assist this the surface unit is placed in a scan whereby the noise observed on each channel can be compared.

3.2 Scenario 2 - Surface to Underwater On Offshore Platforms.

The acoustic situation is shown in Figure 6. While apparently quite similar to the first scenario there are three significant changes. First the depth is generally limited to 500M. Secondly there would be a significantly higher noise level from the platform itself and third the platform itself could cause additional reflected acoustic paths which are very similar in form and path length to the direct path.

The limited path lengths allow much higher frequencies to be used with typical values from 60 to 140 KHz. Directional transducers can be used at both ends since, in general, data gathering packages would be fixed to the platform. The principal paths are the same as scenario 1, namely the direct path(s), the surface reflected path and the triple pass with surface and bottom bounce. These are shown in Figure 7.

Data from the LUNA B platform off Crotone, Italy (Reference 6) shows that the noise from the platform is again really only significant below about 40 KHz. Therefore since only higher frequencies than this are anticipated there is no significance in the local noise. The data is presented in figure 8.

The presence of local reflections is of great importance. Using the FM signal described earlier the effect on the received signal is to add or subtract to the signal depending on the frequency and path length differential. The result is a skewing of the signal as shown in figure 33. In the hardware section the development of a digital AGC will be described. For the purposes here it is important to note that if the AGC is used to deskew the signal the result is excessive induced AGC noise in the decoder and sharply reduced performance of the system. The deskewing method which does work well uses additional analogue AGC sections within the transverse filter array.

The results of the analysis is similar to scenario 1. The frequency chosen is as high as possible. The decoder can be chosen to be the hard limiter or the transverse filter array. In the second case the data is limited to about 3 Kbaud. With the hard limiter higher data rates are possible provided that the skewed signal can be tolerated or deskewing circuitry added. As before data rates may be limited by bit jitter problems, however.

An additional problem may occur. If a portion of the structure is below the underwater unit and has significant sonar cross section there could be a much higher value for the triple pass multipath. In a limiting case the signal could be assumed to have no bottom loss and this is shown in Figure 7 as H. This could also apply to a very hard bottom such as rock. Note that this is a very stringent assumption since the "perfect" reflector also must lie just beneath the underwater unit. In practice the underwater units would probably be located off the bottom.

The significance of such a strong multipath would dictate the use of the filter array decoder in this instance. Indeed for the offshore platform problem it is probably better to consider the filter array decoder as the primary choice and the hard limiter only used when absence of bit jitter and multipath problems have been observed from the analysis.

Hence the adaptive criteria become:

1. Determine absence/presence of significant multipath.

2. Measure skew or include deskew in design.

3. Measure bit jitter.

4. Select decoder.

5. Select highest frequency consistent with S/N > 14 dB.

6. Select highest data rate.

If for some reason there is exceptionally high noise at the time that telemetry is needed then an additional 20 dB of margin can be achieved by using the probability decoder. This decision would appear at item 5 above.

The tower problem is a very real application which is currently receiving attention. Because it is relatively simple acoustically the frequency adaptability may not be necessary in many instances but data rate and power level adaptability are.

3.3 Scenario 3 - Communications With a Remote Vehicle From an Offshore Platform.

This scenario is shown in Figure 9. The physical parameters can vary considerably with significant differences in the acoustics. For purposes of the calculations the depth is assumed constant. The vehicle can vary in location both in depth and distance from the platform (range). For purposes of this analysis the depth is fixed at 150M and the vehicle is at three depths 50, 100 and 140M. The range or distance from the platform is limited to 2 KM.

The temperature profile with depth becomes a significant factor for this problem. Two profiles were assumed. Both have a temperature declining with depth above 90M with constant temperature below this depth. In one case there is a constant temperature layer above 30M. This is shown in terms of the sound velocity profile in Figure 9. The results for the two profiles are similar in nature but the effective reach of the individual ray families is affected considerably.

The surface transucer is assumed to be located 10M below the surface and is initially assumed to be omnidirectional. The acoustic rays that intersect both this transducer and the vehicle can be denoted as

R	Direct ray
S	Single reflection from surface
B	Single reflection from bottom
SB	Reflection from surface and bottom

Many other ray families exist but all involve considerably larger path lengths and hence attenuation. Examples of the ray paths are shown in Figure 10.

Figure 11 summarizes the four families in terms of initial angle and range for 100M depth. Figure 12 shows the signal strengths as a function of range for different frequencies and Figure 13 shows the effect of varying frequency at 1KM, within the reach of the R and S rays. Figure 14 shows the signal strength at 2 KM where the B ray will be dominant. Several comments can be made.

For the specific vehicle and water depths the R/S rays extend out to about 1 KM and the B and SB rays to 2 KM. Beyond 2 KM greater numbers of surface and bottom reflection will be necessary. Within 1 KM frequencies above 50 KHz are needed. As the range is increased a point is reached where the R and S rays quickly disappear and, since the B and SB rays have larger attenuation (from bottom losses) The maximum frequency drops dramatically. How much drop will depend on the nature of the bottom and on the sound velocity profile. An example of this change is shown in figure 15 where the frequency for 10 dB S/N is calculated at different ranges. Clearly as the vehicle changes position frequency adjustments can be very effective.

At short ranges the problem becomes similar to the previous scenarios and a directional transducer is effective at reducing the first multipath signal strength (see figure 7).

Therefore the adaptive criteria becomes:

1. Select maximum signal strength

2. Determine levels of signal, noise and multipath

3. Increase frequency until the S/N and S/M margins are equal

4. Select transverse filter array

5. If the range is short (e.g. less than 5 times depth) use directional transducer.

The frequency capability of the system will depend on the scale of the problem. As an example for communication up to 2 KM frequencies from 20 to 100 KHz will be necessary.

A particular problem exists if the vehicle is at the edge of R and S rays. Time varying profiles will cause the signal strength to fluctuate strongly as the differing ray families each become prime. The system would need to recognize the situation so that parameters are not optimized for one family and then used for a different family. In the example of Figure 13 when the range exceeds 1 KM the prime switches from R to S rays and at 1.4 KM the prime becomes the B rays. At each change the S/N and S/M differences are also changed.

The system to be described later will use a digital AGC and therefore the signal level changes associated with ray family changes will not be apparent if a transmission is in progress. Therefore some continuing monitoring of data error is important.

Therefore to the above criteria for parameter selection must be added a new one:

1. Monitor data error rate continually.

2. If error rate rises reinitiate the process.

3.4 Scenario 4 - Communications From A Platform To Nearby Pipeline Locations.

This scenario refers to a real situation on a platform off Crotone, Italy. Therefore actual engineering parameters were used in the analysis of this scenario used. Figure 16 illustrates the situation. The platform is located in 70M of water and pipelines extend offshore into deeper water. The slope is curved and reaches 450M water depth approximately 7 KM from the platform

The sound velocity profiles are summarized in three catagories as shown in Figure 17. In a similar form to scenario 3 the profiles have a temperature decline and an isothermal layer below 90M. The surface may be isothermal at times. In addition a warm water layer occasionally intrudes at low levels. The temperature profile lead to acoustic ray paths which curve downward above 90M. Therefore in order to extend each family of rays outward as much as possible the transducer at the platform is located near the sea floor. The pipeline transceivers are also located close to the sea floor.

Examples of the acoustic ray paths are shown in Figure 18. The refracting rays are direct to approximately 1500M with the first bottom reflecting ray reaching down the slope several kilometers. Because of the bottom topography many rays may reach down the slope.

The presence of isothermal water near the surface (Figure 19) changes the location of the rays and extends the reach of the S rays further down the slope. The R rays are not affected significantly.

The occasional deep warm water is very significant since it causes the rays to curve upwards and creates a shadow zone near the ocean floor. In this case communications beyond 3 - 4 KM may be impossible.

Summation of the ray histories is shown in Figure 20, 21 and 22 where the identification of ray families at differing locations is apparent. This allows the identification of the prime and first order multipath rays.

An example of signal strengths at constant frequencies is seen in Figure 23. A frequency of 20 KHz is used. The R rays are prime to 1.8 KM with the S rays prime from 1.8 to 2.8 KM. Beyond that the RBR rays will be prime.

Isolation of the prime and multipath rays at short ranges can be achieved by a directional transducer as can be observed by Figure 20. Below 500M the initial rays are 20 degrees apart and some 5 dB attenuation of the S rays can be realized.

The adaptive process is somewhat simpler since it depends less on multipath problems than on signal to noise problems. Since the presence of specific ray families depends on the thermal environment and as ray families change significant signal strength changes occur monitoring of the signal to noise is important.

A significant factor is the variability of the noise as the distance from the platform increases. The transmitter link must recognize the noise at the other receiver and therefore this information must be included in the conversation.

The adaptive criteria are:

1. Measure signal and noise.
2. Identify downlink noise.
3. Increase frequency until limiting S/N (14 dB).
4. Measure multipath and if critical increase frequency further.
5. Adjust data rate for bit jitter.
6. Include signal/noise in transmission.

3.5 Summary

The scenario analyses are not meant to provide specific operating parameter values since they depend greatly on the assumptions of geometry, topography, thermal profiles, etc. However the trends are typical so that parameters are identified.

It is possible to prepare a list of procedure and system requirements for the adaptive software and hardware. As an example the initial sequence becomes:

Measure ambient noise (all channels)
Estimate range
Calculate frequency

Set full power
Set low data rate
Select transverse filter decoder
Transmit initial request identifying:
  Frequency
  Data rate
  Power level
  Local noise

and the first receive and answer operation is:

Scan channels to identify input
Determine data rate and bit coherence
Confirm incoming data
Identify parameters in use
Measure S/M
Measure S/N
Identify if probability decoder possible
Select decoder
Increase frequency if noise allows
Increase data rate to maximum allowed (from bit jitter measuement)
Reduce power if maximum frequency possible gives excess S/N
Retransmit with above parameters

Subsequent transmissions reconfigure as necessary. The basic goal is to reach optimum by the fourth transmission. It should be noted that the sequence above increases frequency before it reduces power. This procedure must be modified if the filter array decoder is selected, the frequency exceeds 60 KHz and there is sufficient S/M margin. Since the maximum data rate has already been reached power reduction becomes a primary consideration.

With these procedures identified the development of hardware and software are possible. This development produced the tools whereby the concept of adapable acoustic communications could be evaluated in a set of laboratory and ocean experiments.

4. HARDWARE IMPLEMENTATION OF ADAPTIVE FEATURES.

The basic system will use microprocessors to implement the functions of acoustic analysis, signal generation, detection and decoding, selection of operating parameters etc. There are several important hardware components which are needed to allow the system design to be completed.

The primary design element is the transverse filter array since this makes the selective decoder possible and allows for easy changes in acoustic frequencies. Other acoustic signal processing elements are matched to the filter characteristics and include the signal synthesizer, a digital AGC, the deskewing circuits and the presence detectors used to reconstruct the coded digital data.

The digital processing involves the data coder and the data decoder. The probability decoder is an adjunct part of the design.

Transducers suitable for acoustic signal generation and reception are also elements of the design.

4.1 Transverse filter array

The transverse filter element is the Reticon 5602-3 (or 5602-3A) devices. These are bucket brigade devices where the signal is clocked along a delay line according to an externally applied clock signal. Weighted summations of the signals in the line is made according to masked creation of split electrodes of different relative sizes.

The result is a narrow band filter with bandwidth equal to 22% of the center frequency and with edge roll off typically of 150 dB/octave. The center frequency is 1/16 of the applied clock frequency and is variable from 250 Hz to 250 KHz.

By cascading filters and offsetting their respective clock rates very narrow band filters can be generated. In particular examine the arrangement of figure 24. Three filters are used with the output of one routed to two others. The clocks are in the ratio 5:6:7 and the resulting outputs are shown in figure 25.

The resulting array has two filter channels which can discriminate two signals of ratio 11/13 with 20 dB isolation.

Hence if a signal which is modulated between the two frequencies enters the array, the array serves to separate the two so that the outputs depend on the presence of either signal.

In addition to the steady response of the array the dynamic response of the filters is of interest since the maximum modulated data rate will depend on the filter response. The filter can be regarded as similar to a high Q analogue filter which "rings" on application of a pulsed signal. Figure 26 shows the effect of inputting a pulsed signal into a single filter. There is a period before which a sufficient number of buckets in the delay line have suitable data. Similarly when the input signal ceases there is still some signal present to produce an output. These reaction times are shown in figure 27. Clearly the filter response will result in a very specific maximum data rate. Since the actual signal is not pulsed but is swept from a signal within the filter passband to one outside it a somewhat different response is observed. There is indeed an improvement in the effective reaction times.

The maximum data rate through a three filter array is shown in figure 28. The data rate is proportional to center frequency up to a data rate of about 3 Kbaud and then becomes nearly constant. Hence the filter decoder has a very specific upper limit to data rate.

4.2 Signal Synthesizer.

The fact that the filter clocks and the acoustic frequencies are exactly proportional suggest the use of a single synthesizer to create all the frequencies. Using a microcomputer to set the synthesizer constant allows easy selection of the acoustic channel. This is illustrated in figure 29. The frequencies are set by selection of a single constant J. In practice J is a two digit number allowing 100 frequencies. For most of the systems constructed by the author a set number of channels is used (either 4 or 8) and a table of values for J sets the frequencies.

In the circuit of figure 29 a reference signal is selected by wire jumper. The basic frequency (denoted as Fx) is generated by the phase locked loop and divider and is simply J X F(ref). Dividers of 5,6 and 7 produce subsets of the digital filter clocks. These signals are multiplied by eight to produce the clock drive frequencies. Using the CMOS phase locked loop 4046 at 15V it is possible to generate clock frequencies over 2 MHz for acoustic signals of 130 KHz.

The acoustic signal is first generated as clock pulses using a divider of 11 or 13 with the value selected by the PWM data. (It is usual to use the lower frequency for the high logic level of the data since the onset of the high is used to regenerate the data clock.) The clock pulses are inputted to a circuit which couples a phase comparator to a sine/square function generator. The square waves are used by the comparator and the sine waves are the acoustic signal source.

Figure 30 shows a short computer calculation of typical frequencies generated by such a synthesizer. It is useful to select data channels which are excluded by the transverse filter set of the adjacent channels. This implies a ratio of at least 1.29/1 from the center frequency of one channel to the next. Thus a set of four channels with frequencies set as close as possible the overall frequency ratio (from the lower frequency of the lowest to the higher frequency of the highest) will be about 3.27/1. For an eight channel set it is about 9/1. This imposes a significant requirement on the transducer.

Thus it can be seen that the simple synthesizer/filter array combination will allow acoustic frequency selection over a wide range with only software changes in the controlling microcomputer.

4.3 Filter Array Data Reconstruction.

The output of the filter array is the signal divided into the high and low frequencies. If the input to the array is normalized so that the mean peak amplitude is maintained constant (at 0 dBV) then the filters are operating at their optimum condition and the output levels will also be in effect normalized. The presence of each signal is detected by a comparator which will produce pulses at the input frequency if the level exceeds a certain value. This value is determined by a circuit which uses the peak value and the value when the signal is "absent".

This "absent" value will depend on the level of multipath signal. As the multipath signal is increased its contribution to the filter output will also increase reducing the margin for the comparator. To improve the effectiveness a capacitor can be used to respond to the integral of the signal presence.

In practice only the onset of the presence of each signal is used. This triggers a multivibrator (retriggerable) with time constant chosen so that in normal operation the output will stay on if the signal is present. The time constant is thus limiting on the data rate since the pulse length must be sufficiently short to allow resetting during the data absence. A longer pulse duration is in effect a filter which improves the signal to noise capability of the detector. In most applications the range of data rate is a compromise between speed and noise rejection.

Since the maximum data rate is a function of acoustic frequency it is advantageous to adjust the pulse length to the acoustic channel. The pulse length is set to just allow the maximum rate for the channel selected. The usual method is to switch the resistor/capacitor values in the circuit according to the selected channel number(see figure 31).

Figure 32 demonstrates the signals within the decoder. The outputs of the pulse generators are used to trigger a flip-flop and the output of this becomes the serial coded data.

4.4 Deskewing Of The Filter Array Outputs.

If a near reflection of the acoustic signal occurs the received signal may be skewed. This would be apparent in the vicinity of a tower or other similar structure and where a directional transducer cannot isolate the signal. If an insufficiently directional transducer were used near the ocean surface a similer situation would occur. This skewing is demonstrated in figure 33. In this case since the signal input to the filtar array is normalized with respect to the larger of the two signals the other frequency would be at reduced level entering the presence detector.

To recover the signal and allow decoding additional analogue AGC circuits are used. This is shown in figure 34. The fixed gain amplifier which serves to recover the filter insertion loss is replaced with a variable gain amplifier whose gain is set according to the peak output level. Note that since one of the signals must be normalized and one at reduced level (due to the AGC operation) these analogue variable gain amplifiers need only operate over a limited range and hence can have quite short time constants.

The result is that a signal which is skewed as much as 20dB can be reconstructed so that the final skewing is only 1 dB. This small residual skew does not affect the system decoders and error free data can result.

4.5 Automatic Gain Control.

The performance of the filter array depends on the normalization of the input signal. An AGC circuit is needed. However, in addition to the basic function of variable amplification, an output which measures input signal level is also required. This is to provide inputs to the acoustic analysis performed by the data computer.

It is clear that the input signal can vary over a large range of amplitudes. At least 60 dB of AGC control is needed. Therefore a digital AGC amplifier is used. This is controlled by an eight bit microprocessor.

Figure 35 shows the basic circuit. Three stages of amplification are employed. The first two provide coarse amplification changes in steps of 8 dB while the last amplifier provides 1 dB steps. Switched resistor circuits are used. In practical examples .1 dB accuracy is achieved and is adequate for the purposes of the design.

The final circuit can provide 64 dB of dynamic range from -8 dB to +56 dB gains in 1 dB steps. The amplifier is effective up to 140 KHz. The level detector circuit uses capacitor filters to reduce the response time of the circuit. This was found to be the most effective arrangement during testing. With no filtering the circuit can provide as much as 120 dB/millisecond response.

The AGC circuitry is usually accompanied by a preamplifier and switched analogue filters. These filters are tuned to the acoustic frequencies for each channel. They serve the purpose of splitting the input signals so that the noise distribution can be determined. The preamplifiers and filters provide 64 dB of gain. Hence the total gain of the analogue circuitry is 120 dB with a nominal output of 0 dB. The effective sensitivity of the system should be -120 dBV or 1 microvolt. While the preamplifier can certainly be sufficiently quiet it is necessary to be be very careful in circuit layout to prevent noise pickup from the digital circuitry.

With signals larger than -64 dB the AGC circuit saturates. The output signal is then limited by the circuitry to 5V p/p or about 5 dB. The signal will, of course, be clipped so as to produce harmonics. Provided that the digital filters bias levels are maintained the filter array will continue to decode the data up to very high signal levels. The total dynamic range measured on a recent system was above 80 dB.

The measurement of the signal strength is easily performed by reading the value of the digital gain word. This 6 bit word is inverted in the software to provide the correct data. The level of the multipath signal must be obtained by measurements within the digital filter array. The comparator circuit provides an analogue signal sensitive to the multipath presence and this is measured by the data computer by a simple A/D circuit. To calibrate the measurement so obtained a multipath generator with known levels is used. This is described later.

4.6 Data Generator.

Creation of the pulse width modulated data is accomplished by the method of figure 36. The clock Fx is used so that the data is proportional to the acoustic signal. (With acoustic frequencies above 60KHz a constant reference clock can be used). This is divided by a constant inputted from the data computer. The resulting variable clock sets the data rate.

Parallel data from the data computer is inputted to a FIFO array and the outputs of this FIFO array is converted to NRZ data. This NRZ data is sampled by the modulator so that the PWM data is created. The actual proportions of the data can be varied at will. For most of the applications the data has a 2/2/2 or 2/3/2 distribution.

This means that the data portion of the bit can be either equal to the leading and trailing portions or 50% longer. The longer format gives slightly slower data rates but improved error rates on decoding.

4.7 Data Decoding.

Decoding of the data stream would be straightforward except that the data rate is unknown and the bits are not steady. The data decoder must be able to detect the data rate and decode it. Use of standard phase locked loops is not practical under such conditions.

The data decoder is shown in figure 37. Its operation is shown in figure 38. The bit start is detected and a counter starts. At the completion of the bit (start of the next bit) the count is recorded. Thus the count measures the bit duration (bit rate) and the consistency of successive bits determines the bit jitter. Observing bits whose variation is less than a preset standard is denoted as bit coherence and this observation is a major step in data conversion.

The decoder mode can be changed so that the counter halts at the termination of the high level of the bit. This count duration can be compared with some computed average value to determine the criteria for defining the bits as "one" or "zero". This determinant can be one half of the bit duration. It is better, however, to use a value which is the average of the "one" duration and the "zero" duration.

A third mode is selected in the decoder and the determinant inputted to one side of a comparator. The bits are then converted from PWM to NRZ data by the comparator and the result clocked into a shift register. From signal developed in the circuitry the data is read from the shift register as parallel words to complete the data conversion.

Data error correction codes can be added to the process by software but it is useful occasionally to use parity bits within the data stream to identify errors. This is easily added to the system with hardware. In practice the data format can also be easily changed. Most of the applications to data have used 12 bit data embedded into 16 bit words.

It is obvious that the data rates generated and decoded by the above circuitry can vary over a wide range. The lowest data rate is unlimited but the counter must not overflow. Since a 12 bit counter is used the slowest data is Fc/4096 where Fc is the clock rate. The highest data rate practical is Fc/16 allowing for identifiable counts and high bit jitter. Fc is however selectable. Actual bit rates used have varied from 50 Baud to 50 KBaud. The filters limit the maximum rate to 3 KBaud,however.

4.8 Probability Decoder.

There are occasions where the signal received is so low that the presence detector will not work effectively. This will occur if the signal to noise is less than about +3 dB. Here the noise is measured at the input of the first transverse filter. It is possible to extend the system capability to about -4 dB by using an alternate decoder.

The output of the first transverse filter is inputted to the circuit shown in figure 39. A phase locked loop is used to demodulate the acoustic signal. A second phase locked loop detects the data rate. The signal itself is amplified and a comparator used to limit the signal. Using the data PLL as a reference a count is kept of the period when the signal is above the comparator reference and a second count of when it is below. At the completion of the bit the two counts are compared and the larger count determines the probable value of the bit.

The output of the system is NRZ data and a data clock and these are made available as input to the data decoder for parallel conversion.

While the decoder works very well with very noisy signals it is limited in that the PLLs must be fixed or selected as preset values and there is a loss of flexibility in the selection of acoustic frequency and data rates.

4.9. Hardware Summary.

It is clear that construction of acoustic transceivers around the above design elements allows operation at a wide variation of acoustic frequencies and data rates. Power level variations of the transmitted signal are of course easily accomplished.

Systems built to data use acoustic frequencies from 8 KHz to 140 KHz. Data rates have varied from 50 Baud to 3 Kbaud. Power levels depend on the transducer but signals as high as 170 dB re 1 uPa at 1M for the transmitter have been used with typically 10 dB power steps in attenuation.

Transducers used have varied considerably. One of the more useful uses cylindrical elements combined so that the active surface is a cylinder 1.5 inches in diameter and 2 inches long. The elements are wired in series so that high sensitivity in receiving is obtained. The basic element has been measured at -177 dB re 1 uPa output over a wide frequency range. Transmitted sensitivies are correspondingly low peaking at 110 dB re 1 uPa at 1M at 30 KHz. The same transducers have been used in experiments up to 140 KHz.

When applicable the cylindrical transducer element has been used with a conical reflector. Measurements made on this show a stong directionality near the resonance (30KHz). Quite useful front to back ratios are achieved over a range of about 2/1 in frequency (see figure 40).

5. SOFTWARE IMPLEMENTATION OF ADAPTIVE FEATURES INTO SYSTEM

The decision processes described earlier involve making several measurements during the acoustic telemetry and from those measurements defining a set of improved characteristics for an acoustic reply. There are a number of elements in the software process which are keys to the success of the operation. These are:

1. Creation of a message which can provide sufficient information for analysis.

2. Use of a conversation between transceivers which improves the telemetry link.

3. Use of a set of decision making steps to identify incoming data stream and to enable decoding.

Of these the third has proven the most difficult. The incoming data has unknown frequency, unknown data rate and unknown strength. It is necessary to recognize an incoming transmission and to seperate it from noises in the ocean. Moreover since the recognition of the incoming transmission takes several steps it is important to develop a procedure which can trigger out of the process when it determines that a false step has been taken.

The message sent includes the following:

Precurser of alternate 1/0 bits.
Synch word to start each frame
Channel word describing system parameters and message intent.
Signal and noise of last received message.
Redundancy of previous two items.
Repeated checksum words.

Thus a frame of data is constructed with these elements. This frame is used to initialize a conversation and to terminate a series of data frames. It includes sufficient information for the reception of the link, for the analysis of the link and for the determination of the reply characteristics.

The precurser is a string of alternate ones and zeros. The length is sufficient to enable the receiver to identify the channel, determine the bit rate, set the discriminator and check the validity of the incoming stream.

The synch word (3465 octal) is a constant bit pattern which identifies the start of the frame and which distinguishes thé word location within the bit stream.

The channel word contains the transmitting parameters and the intent of the message in the form

C2,C1,C0,X,P1,P2,M1,M2,R3,R2,R1,R0

Where CN is the three bit channel number (two for four frequency systems) and identifies the acoustic frequencies.

X is a single bit instruction usually reserved for command of an underwater release or other special instruction.

PN is the two bit power level

MN is the message intent

RN is the dividend in the data rate equation.

The message intent can be used in many ways. In the experiments made so far the designations are:

MN=0 not used

MN=1 request data

MN=2 conversation reply not including data

MN=3 conversation reply including data

Recognition of the channel word is sufficient to confirm the transmitting parameters and the intent of the message. Except for MN=3 which terminates the process this recognition always initiates a reply.

The signal noise word is a twelve bit word where the higher order 6 bits are the signal level from the AGC and the lower 6 bits are the noise level from the AGC measured just before the incoming message.

The check sum words are the accumulated additions of all data words transmitted at any instant with the lower twelve bits only retained.

The conversation between transceivers starts with the manual selection of parameters and a request for data with MN=1. Since the initial contact may be off the ideal conditions the probability of error is high. Therefore the usual format is a sequence of repeated frames of the form

synch
channel word
signal/noise (signal =0)
checksum

with the last three words repeated 3 times to form a 10 word frame. Repetition of the frame includes the precurser and the frame is repeated a number of times according to the data rate. The criteria is that at least 5 cycles are made and at least 10 seconds of transmission occur.

This gives the receiver several opportunities to receive sufficient information and also include a number of false starts. The reception of a message even one with large errors requires only a single set of four words and from the total frame a value on the error rate can be calculated.

After data reception is complete and an analysis made of the conditions a new set of value is made using the criteria identified in the scenario analyzes. After a delay a reply is sent which may be MN=2 or MN=3. If MN=2 the message takes the above format and if MN=3 the message takes the form

precurser
synch
channel word
signal/noise
checksum
number frames data
checksum
number frames data
checksum
number frames data
checksum
precurser
synch
8 data words
checksum
precurser
etc.
frame terminator

where the frame terminator is the same form as the initial transmission. The precurser between frames is 32 bits of ones/zeros or 2 16 bit words and allows time for data handling. Note that all frames are the same length which simplifies data analysis.

The incoming data stream has a known format and a highly redundant content. The receiving unit must recognize the data as a message, interpret the message, calculate new parameters and transmit an answer. Since the incoming message has an unknown frequency and since the receiver can only receive on one channel at a time it is necessary for the receiver to scan the channels sequentially. The receiver first performs a preliminary scan which it interprets as noise. The scan continues with a second set of signal amplitudes and a comparison with the previous set is made.

In practice it is necessary to perform two scans since a signal can commence anywhere in the cycle. To speed up the process a fast scan is repeated until an increase is seen on any channel. If no increase is observed then the noise is updated. An onset value of 4 dB has been selected for the signal recognition. Thus if the new signal is less than 4 dB more than the old value the noise is updated. With a signal above this level a slow scan is performed. For the systems built to date a fast scan is 20msec per channel and a slow scan is 50 msec per channel.

The slow scan has the function of selecting the channel with the largest gain and determining this as the probable channel. Since this may be erroneous a second probability is also selected.

The system now activates the transverse filters on the selected channel and looks for incoming bits that are coherent. A number of approaches has been used for this step. They include finding five random bits in a set of eight which are coherent or finding three consecutive bits coherent in three tries. This latter is now used.

If coherence is not found then the second probable channel is tried and if again unsuccessful the system reverts to scan.

If coherence is found then the one/zero measurement is attempted. The incoming data has a one/zero preamble and to identify the bit discriminator four consecutive bits sequentially greater and less than one half the bit duration are sought in a eight bit sequence. Again if not found the system reverts to the second channel etc.

With the discriminator set three consecutive words of value 2525 octal are sought with similar time constraints. Kick out is to the second channel.

The system now looks for the synch word and when found sets the word identifier and reads the next 9 words into a register. It then performs an evaluation of validity of the message. This validation takes several steps.

1. first checksum correct?
2. channel word consistent?
3. channel word repeated?
4. message intent?

If data is identified as incoming the system will receive and record data as eight word blocks (in a ten word frame) until complete. If the frame was an M1 or M2 message then an analysis is performed on the acoustics. The multipath level is measured immediately and the signal strength and last measured noise recorded. In addition the whole ten word frame is analyzed for bit errors. The system can now evaluate the link and calculate new parameters.

The calculation of bit jitter and its contribution to bit errors can be made directly from the bit measurements. It is necessary to set some upper value for the acceptable jitter. Even though the bits can be decoded with up to $\pm$ 12.5% jitter a limit of half this value for eight consecutive bits has been determined to be more useful and the data rate can be changed to give this value.

Calculation of the new frequency is more involved and will depend on the scenario. Hence different programs have been used for differing scenarios. Indeed for some scenarios frequency changes are not made with software but are preset before installation. Power level and data rate are retained as variables and these systems are partially adaptive.

There are clearly several decision or go/no go points in the software and the system is quite critical in recognizing a message. Originally it was felt that bit coherence was an adequate identifier in itself. During sea tests, however, bit coherence from ocean noise was a regular occurrence and not a critical step. The key identifier now appears to be the recognition of the 2525 octal word after the discriminator has been set.

A second important point is that if the oceanographic noise is above the sensitivity level of the system which is common at acoustic frequencies below about 50 KHz the noise will appear as random bits in the discriminator. If the ocean is quiet no transitions will appear (very long bits!). Therefore a maximum bit duration must be included in the software with suitable kick out when applicable.

There have been no oceanographic experiments with a system with switchable decoders. The criteria for the switch is known but no specific software developments have been made. The decoders have been tested in the laboratory and can be incorporated in systems where suitable.

6. SYSTEM INTEGRATION

The features of the adaptive telemetry system have been described in earlier chapters. These must be integrated into a working system and packaged as telemetry transceivers for practical use. Each station will consist of a transceiver with transmitting and receiving capability.

Several key design requirements are identified for the transceivers. For ease in design and fabrication the two transceivers should share common design features. (The packaging of the two is usually different.) Hence the choice of microprocessors must be made with all engineering features in mind.

At least one of the two transceivers will usually be located under water and frequently remain in position for a long period of time. Therefore one key requirement is a low power system. For this reason an all CMOS design was chosen. Other benefits of this choice is a system which is tolerant to a wide supply voltage variation allowing a free choice of battery types and eliminating voltage regulators. A second benefit is the low power circuitry is easier to isolate and reduce noise and this is important in the design of the low signal level acoustic amplifiers.

At the time this design work was initiated two CMOS microprocessors existed. These were the Intersil 6100 12 bit device which was an LSI implementation of the venerable PDP 8 minicomputer. The second was the RCA 1802 8 bit device. Both of these are true static CMOS devices. Since that time many other CMOS microprocessors have been designed. Some of these are true devices while others are hybrid with CMOS sections as only part of the design. In fact the ease of programming of the 6100 still makes it or its newer brother the 6120 a good choice.

The 6100 was chosen for the main function CPU because its 12 bit format was a natural for engineering parameters and because the elegant instruction set allows easy programming and debugging. The design of the system also incorporated a direct read/write capability from a PDP 8 and this in turn provided a sophisticated development system for immediate operation.

A key design feature is the use of a digital AGC. For this operation the 1802 was chosen because the 8 bits provided adequate single word resolution and the internal registers allowed the elimination of external RAM memory.

Since long period operation is a natural design requirement several basic design features were incorporated to reduce energy consumption. These were:

1. A variable speed microprocessor clock controlled by the system itself.

2. The use of a segmented bus to allow power switching of sections of the computers.

The basic system design is shown in figure 41. The functions are separated into two groups each involving a microcomputer with control and data buses connecting the two. The data computer is the primary controller with the acoustic computer acting as a slave processor. The bus on the data computer is divided into a maximum of three segments. The bus on the acoustic computer is not segmented but the functions are power strobed as required.

6.1 The data computer.

The data computer must perform the following functions:

1. Acquisition of engineering data external to system.
2. Storage of data.
3. Formatting of data for transmission.

4. Control of transmitting operation.
5. Control of receiving operation.
6. Decoding of the data stream.

7. Measurement of the acoustic parameters.
8. Calculation of the acoustic performance.
9. Calculation of revised acoustic parameters.

10. Determination of reply if necessary.
11. Interaction with other computers or (in the case of the deck unit) an operator.

In most of the operating scenarios that have received attention to date one of the transceivers has been an underwater unit and the second has been a manual or computer operated deck unit. Operations have been initiated by the deck unit and the initial set of parameters have been chosen either from knowledge of the acoustic situation or as part of an experimental sequence.

The data computer is shown in more detail in figure 42. The components are fairly straightforward. They are be divided into several power segments. The functional elements are:

P0 Real time clock.
Non volatile memory (carry over data)
Interval clock.
Start up circuitry.

P1 CPU
Program memory
Bus Isolation Circuitry.
Data memory.

P2 External interface (to manual or computer control)
Acoustic data formatter and control
Acoustic data receiver and decoder.

The three power functions operate as required. Typical operation is as follows. The real time clock operates continuously (but on reduced voltage). The interval clock is set by the CPU before it becomes quiesent. At some preset interval the CPU and P1 is initiated. In usual operation the interval is set to the start of the next minute (seconds = zero). The CPU can then read the clock and examine its timetable for the operating cycle. For example the underwater unit can be programmed to listen for acoustic signals from the deck unit over selected periods. All calender functions are used including the day of the week. A manual override is used for manual input to the deck unit.

With P1 active and P0 redefined to a higher value the CPU loads subroutine entries and continues according to its schedule. If no operation is indicated the CPU can reset the interval clock and become quiesent. If outside activity is indicated the CPU activates P2. All other power controls P3, P4 and P5 are set through the P2 section.

The microprocessor clock is powered by P1 and derived from the real time clock oscillator. Frequency multiplication is perfomed using a phase locked loop and divider. The clock selection will depend on function but typical values are:

DMA from external development system	- 131 KHz
Transmit cycle	- 524 KHz
Receive cycle	- 1048 KHz
Compute cycle	- 2096 KHz

Figure 43 shows the clock (P0) functions. The real time clock is a OKI MM5824 chip. Other elements are standard 4000 series CMOS devices.

Figure 44 shows the P1 grouping. The program is stored in

6653 CMOS ePROMS. The normal operation uses a total of 4K of CMOS memory with optional division of PROM and RAM in 1K intervals. For most design functions to date only 2K of PROM are needed.

For one special application an extensive memory was needed to store data over a period of time before telemetry. For this purpose an additional memory bus was used with CMOS ram memory added in 8K word increments. 256K words were added although this can be enlarged if necessary. The 6100 instruction set was augmented with additional I/O instructions for this purpose.

The bus isolation circuits are shown in figure 45. Note that the isolation is in two parts, power and signal since it is important to stabilize the power before signal application to avoid latch up of the circuitry. In all four examples of acoustic telemetry systems have been built with differing levels of adaptability. Each one has used a different configuration. It is possible however to summarize them as to power consumption of the underwater units. Typically the power consumptions and comments on duty cycle are:

P0	100 uA	Always on
P1	10 mA	1 second /minute
P2	25 mA	As needed.

6.2 The acoustic computer.

As noted earlier this uses an 1802 microprocessor. The computer is under control of the data computer and performs the following functions:

1. Setting synthesizer constants.
2. Operating AGC
3. Outputting gain (signal level)
4. Outputting multipath measurement.

The bus is not segmented as it is for the data computer but sections of the system operate under differing power sources. Figure 46. shows the computer. The functional elements are:

P3. CPU
Program memory
Synthesizer
AGC and analogue receiver elements

P4. Receiver transverse filter elements.

P5. Transmitter function generator
Transmitter power amplifier.

Most of the important elements have been described earlier. The program for the 1802 resides in a 6654 CMOS PROM which also includes constants for the synthesizer. The CPU is initialized by the data computer and upon initialization sets the synthesizer constants from data in the PROM. Changing frequency sets is as easy as changing this PROM.

During transmitting periods the CPU holds the AGC at the minimum gain condition. During receiving periods the CPU operates the AGC and the value of the gain is available for the data computer at the interface.

The multipath signal value is determined by the circuit of figure 47. Since the "signal" is normalized the multipath is measured when the signal is absent. To achieve this the signal is sampled and the value determined by an A/D converter. The value is made available as an alternate data set on the AGC gain lines and is selected by the data computer.

Note that this multipath measurement is only used on the fully adaptive system. Systems which adapt data rate and power level only do not use the multipath data.

The power amplifier circuit uses two RCA 2002 devices as push pull drivers for a toroidal transformer. These devices are normally used for automobile radios but have been used up to 140 KHz in some of the systems. Some care to prevent oscillations of the amplifiers is necessary.

The acoustic input signal from the transducer first enters a preamplifier. In the underwater units the transducer is located very close to the transceiver and the preamplifier is located within the SPW or transmitter power board. In the deck unit the transducer is often located as much as 150 M from the transceiver and the preamplifier is located within the transducer housing with power and signals present in the connecting cable. It has been possible to use a single conductor cable with double armoring in all applications.

The preamplifier design is straightforward. It uses a 2N4868A FET transistor as the front end. With this the front end electronic noise is always less than 1 uV. provided that care is taken in the layout of the components.

The basic preamplifier has an extremely wide frequency response, from D.C. to several MHz. For most acoustic applications this is far in excess of the required response and analogue filtering is usually included in the design.

6.4 System packaging.

All systems built to date have used different packaging for the deck unit. The underwater unit packaging has stayed the same. An example of a deck unit is shown in figure 48. A manual keyboard is integrated into the system to allow manual operation of the deck unit. The keyboard can also connect to the underwater unit to set the clock and to allow system checkout. An additional interface allows the deck unit to communicate data and commands to and from another computer. In one of the systems for example this computer is a HP85.

The underwater unit is housed in a pressure canister. The electronic assembly is shown in figure 49. The frame is very rugged and can withstand extremely rough handling. The design is identical with that used for an air deployed buoy developed for the U.S Navy by the author (Reference 7). It is designed for 100 G for 25 milliseconds but has been tested up to 1000 G for 10 microseconds. No problems with shock damage to the computer have ever been observed.

The transducers used depend on the particular application. For most applications a cylindrical transducer of 1.5 inches diameter and 2 inches length (active element dimensions) has been quite effective. Where directionality is required this has been inserted into a conical reflector (see the sketch in figure 50).

The power for the deck unit has been provided by a single "gel-cel" battery pack of nominal 18 V and 6 AH capacity. This is maintained by a controlled current charger with auto cutoff. The deck unit can therefore operate without external power. The period of operation will depend primarily on the number of transmissions and the power levels but operation over an eight hour day is typical.

Power for the underwater unit has varied but for long period operation a lithium battery pack has been often used. These employ the usual safety design factors and have been trouble free over many years of operation.

In summary of the hardware integration, the basic computer design has now been in use for some five years in various forms. While another microprocessor could will be used in a new design the 6100, like its predecessor, the PDP 8, has proven a good trouble free performer.

7. SYSTEM TESTS

The integrated system has been created, to date, in four forms. Two of these have been very variable in form and have been primarily used for laboratory measurements and local sea trials. One of the systems is being used as a basis for the development of a tower monitoring study in the North Sea. This extension is under the supervision of Det norske Veritas in Oslo, Norway. The fourth system is being integrated into a computer operated test along the sea floor off the coast of Italy by Tecnomare.

Each system represents an improvement with configuration and software changes. The following tests were performed over a period of time with all four systems. The data represents the best examples and in general, correspond to the most recent system.

The tests can logically be separated into two parts, those performed in the laboratory and sea trials. The laboratory tests, made under controlled conditions, give more insight into the potential of the approach while the sea trials were generally more of a demonstrational nature since many environmental factors which influence the results could not be measured at the time.

7.1 Laboratory tests.

The laboratory tests were performed to determine the following information:

1. System power consumption.
2. System sensitivity.
3. Dynamic range.
4. Minimum signal to noise.
5. Minimum signal to multipath.
6. Maximum combined noise and multipath.
7. Maximum data rate.
8. Minimum data rate.
9. Effectiveness of channel selection process.

These tests allow some interpretation of the systems ability to adapt to data rate and power level changes. The tests noted above imply that both the transducer and electronics are involved and in general these were evaluated separately. Hence the transducer data can be used to calculate the system performance but a change in transducer can be made and the characteristics of the new transducer can easily be used to calculate a new system performance.

The tests on power consumption are straightforward and involve monitoring current consumption during various phases of the operation. Since the underwater unit is the more critical with respect to power consumption the data usually refers to this unit. Typical values are:

Mode	Current (mA)	Notes.
0	.1	Idle with clock and interval timer
1	10	Evaluate schedule
2	25-30	Receive (no transverse filters)
3	120	Receive (transverse filters on)
4	200-1000	Transmit (depends on power level)
5	25	Compute (prior to answer)

In fact the clock consumption has been tested to 30 uA at 2.4V but this was found to be unsatisfactory because of problems with start up of mode 1. If the carry over memory is not included then the need for system maintenance requires the P1 power not drop below 2.2 V and the mode 0 consumption rises to about 2 mA. If extended memory for data storage between communications is added then about 1 mA per 8Kwords must be added to the mode 0 level. The deck unit consumes more power since a level converter to RS232C is used for simplification of the data interface to a variety of terminals.

The current consumptions above can be used to calculate system operations if certain aspects of operations and a typical battery pack are assumed. Consider the following:

Battery pack - 30 Lithium D cells (Duracell LO26S) as 4 banks of 5 cells to give 18V when transmitting and 20 V when receiving.

Configuration - 8K words of data storage and carry over RAM

Operation - Preset listen intervals 6 times a day for 30 minute intervals with staggered scan.

Transmissions at varying powers for a total of 240 secs. per day.

This would give over 6 months of operation with ample margin. Note that the receive operation which uses the transverse filters for about 30% of the receive time uses about 70% of the energy available.

Clearly the operation above could be tailored to give longer durations but if the transverse filters were not used or were replaced by CMOS devices the system would be more easily tailored for longer operations.

The tests for the system sensitivity involve the electronics and the transducer characteristics. The electronic system will depend on the preamplifier noise and on any noise pickup from the digital circuitry. Indeed suppression of noise from the digital filters is the most important task in the layout design. For the most significance then the digital filters must be in operation during the tests. Figure 51 shows the test setup. The receiver is programmed to hold in the filter on mode and to interpret the incoming stream of alternate one/zero bits. A count is kept of the errors (defined as any bit not meeting the expected value) in 10Kbits of data. The signal level is increased until the error became non zero and then decreased until the error rate became zero again.

This measurement was made for two systems and for both deck and underwater units. The results are:

Frequency	System T		System E	
	Deck	U/W	Deck	U/W
(KHz)	-----------dBV-------------			
8.3	-116	-118		
12	-116	-118	-108	-110
18	-118	-120	-105	-105
23			-111	-112
27	-123	-124		
30			-115	-116
40			-120	-120
Above 40			-120	-120

Note that except at low frequencies the sensitivity is generally around -120 dB rms or 1 uV. There is also a very definite tendency for the sensitivity to decrease at lower frequencies. It should be observed that the preamplifier has an essentially flat response at all frequencies above 8 KHz.

The maximum signal with which the system decodes "error free" data can be measured the same way. The attenuator is not used and the signal measured as before. The signal can also be easily observed on an oscilloscope.

The maximum signals measured in this case do not have the same consistency as the low level signals. After about 60 dB of increased signal (over the low value) the signal begins to saturate in the last stages of the AGC circuitry. The decoder continues to perform until at least 12 dB of further increase and in some cases for larger signals. (Note that there is no "multipath" signal so that signal to noise/multipath" is very high. The minimum dynamic range is then 72 dB with some cases of dynamic range greater than 80 dB observed.

In summary the dynamic range in terms of the input signal is -120 dBV to -48 dBV. It is useful to relate this to the acoustic signal and this will be done later when the transducer calibrations are shown.

The minimum signal to noise is measured in a similar manner to the sensitivity test. A signal well above the natural noise is used. The setup is shown in figure 52. The noise is injected from a white noise generator. The signal and noise are measured at the input to the transverse filter array by eliminating each in turn. The result is that over most of the frequency range the noise must be at least 3 dB below the signal for error free decoding to occur. Note that the signal to noise at the output of each path through the transverse filter array is larger than 3 dB since the bandpass of the filters is very much smaller than the analogue filters preceeding the AGC. Note also that there was no observable trend of minimum signal to noise with frequency though the measurement was not very precise. The uncertainty of the measurment was at least 2 dB so that -3 to -5 dB would better represent the results.

Measurement of the minimum signal to multipath requires the use of two generated signal displaced in time and of differing amplitude. The method used is shown in figure 53. Two function generators are used and the "data" for the second is merely delayed from the first. The amplitude of the first was adjusted to be well above the natural noise and the second amplitude adjusted until the decoder error rate was non zero. The best case was where the frequencies were in the range (20 - 50 KHz). The minimum signal to "multipath" measured in this manner was about 4 dB. Under some circumstances this was as high as 7 dB.

Note that both the minimum signal to noise and the minimum signal to multipath were measured with excess of the other factor but in a real situation both will be influential. To evaluate the combined effect noise is added to the setup of figure 53. When either the signal to noise or the signal to multipath were less than 10 dB the minimum for the other factor became larger. Hence the results are indicated in crude form in figure 55 and the uncertainty band of at least 2 dB in these measurements must be noted.

The maximum data rate is dependent primarily on the response time of the transverse filters but also slightly on the noise in the system. The rate is measured in a similar manner as before. With the decoder operating a series of tests were performed at increasing data rates. Note that the rate must be increased in steps with the decoder initialized between steps since at the beginning of each sequence the decoder must determine the data rate to set the discriminator. Above 60 KHz the maximum data rate is about 3 KBaud. The results have been indicated earlier in figure 2.

The decoder itself only limits the data rate lower limit when the count overflows. Since the count clock can easily be adjusted this represents no problem. There is however an influence from the presence detectors. To improve signal to noise selectivity a pulse stretcher is used. The length of the pulse is chosen to match the maximum data rate. At excessively slow data rates this is no longer effective and a much higher signal excess is needed to achieve error free data.

The system, when in the fully adaptive mode, scans all channels and determines which channel is being used for the communication. A series of tests were used to evaluate the speed and effectiveness of this selection. The acoustic communication is performed in air over a short path (typically 3-4 M) and with various power levels. The underwater unit is programmed to find the input channel and data rate and to reply with the same parameters. The deck unit then confirms the reply. This test evaluates a very necessary step in the adaptability process.

As noted in the earlier description of the software the scan is performed in two parts starting with a slow scan. This determines the possibility of an incoming message. A fast scan then is performed to select the probable channel. The criteria used for the selection is the gain of the AGC. In the tests noted here the units always selected the correct channel provided the signal was at least 6 dB above the noise. This is in accordance with the criteria established in the software.

In order to relate measurements on the electronic system to acoustic signals it is necessary to input data on the transducer characteristics. The transducers used were not necessarily the best available but were found to be adequate for the purposes of this program.

The underwater transducer uses cylindrical elements of 1.5 inches in diameter. It consists of four segments each 0.5 inches in length. The elements are connented in series to provide a high impedance (low capacitance) device. Since the preamplifier has very high impedance a source resistance is used to reduce the Johnson noise and to keep the lower frequency cutoff close to the lowest channel frequency in use. The sensitivity was measured by using a reference transducer of known receiving sensitivity and using two identical transducers as described above. A series of tests was performed over the side of a small vessel with the two transducers in the test located 1 M apart and 2 Meters below the vessel. Water depths were in excess of 5 meters and a short pulse signal was used to identify any reflections. While a more exact measurement can be performed this method is cheap and quite repetitive.

The resulting transmitting and receiving response is shown in figure 55. Note that the response is essentially flat over a wide frequency range and the receiving response is quite high at -178 dB re 1V at 1 uPa. The system response is then 58 to 130 dB re 1 uPa. The transmitting response is accordingly quite low at 110 dB re 1 uPa at 1 M. This is not really a problem since with good insulation on the transducers and with their high effective impedance they can easily be driven with high signal voltages.

For certain applications a directional transducer is needed. For this purpose a reflector was constructed and added to the cylindrical transducer. A sketch is shown in figure 56. The results of the calibration tests are shown in figures 57 and 58. The resonance of the transducer becomes readily apparent in the configuration and the response peaks around 30 KHz. The directionality also peaks in the range and this transducer will be used primarily where the frequency variations are small.

7.2 Sea Tests.

As mentioned above four different systems have been constructed and, to date, three of these have been employed at sea. The Tecnomare system is currently being integrated into the computer network for the Italian trials.

A summary of most of the sea tests is shown in table 1. The sea demonstrations then have involved three systems with different configurations. Initial tests used a single underwater unit and a small portable "deck" unit located on a small vessel. These tests were performed in the waters immediately adjacent to Miami, Florida. The underwater unit was programmed to transmit on a single preselected channel and the deck unit received and decoded the data. The purpose was to evaluate the telemetry link and the receiver process. The data rate was varied so that the data rate adaptability was always used. The underwater unit was initially bottom located but for subsequent tests it was suspended from a surface float at various depths up to 300 M. The location of the float relative to the vessel was always performed using a loran C to identify its location before and after each test and the same loran C to locate the vessel. This measurement is therefore somewhat crude but adequate for the purpose.

A sketch of the configuration is shown in figure 59. Provided sufficient signal existed the data transfer was quite consistent for most frequencies. There was at times considerable environmentally induced bit jitter which limited the maximum data rate. This was not unexpected since the tests were performed on the edge of the Gulf Stream, a somewhat dynamic environment. In general the range of the experiments was limited due to a strong thermal gradient at this location.

A second set of tests used the same configuration and location but programmed the underwater to use different frequencies. This time there was considerable difficulty in acquiring the correct channel. If the correct channel was acquired then the complete message was usually received. To understand the problem a mode and channel indicator was added to the deck unit and the experiments repeated. The observation was made that the deck unit often "acquired" a non existing channel and did not reenter the scan mode unless manually interrupted.

To evaluate this further the deck unit was reprogrammed to indicate the gain levels in all channels on a continuous basis. it was determined that the noise levels were extremely unsteady and the deck unit evaluated a noise pulse as a signal initiation and when in the evaluate mode the noise was insufficient to allow completion of the rejection process in a suitable time frame. A maximum time allowance for rejection was programmed. The deck unit would now continue to incorrectly identify channels but in most cases return to the scan mode quickly (less than 1 second). Correct channel acquisition now occurred about 50% of the time when the signal was adequate.

At this point the channel evaluation used only a bit coherence criteria. It was determined that this criteria was satisfied even if only oceanographic noise was present. The criteria was extended so that both bit coherence and the existence of the bit clock (that is sequential one zero bits) were necessary. The experiments were repeated and the correct channel acquisition increased to about 75%. The evaluation process had now increased in time and it is believed that the non acquisition events were due to the system not being in scan at the correct time.

The experiments continued with a two way transmissions on fixed frequencies. The underwater unit listened for a transmission on a single frequency and replied on that frequency. Over the short ranges that sufficient signal existed the telemetry occurred for a majority of the time. The underwater unit was now programmed to scan and the two telemetry tests repeated. Correct replies were now occurring about 50% but this increased when the transmission preamble was extended.

A set of tests off Key West allowed longer ranges to be achieved but high bit jitter prevented the higher data rates to be used. It should be noted that another unsuccessful series of tests occurred when the underwater unit suspension cable was found to strum with sufficient amplitude to saturate the underwater premplifier circuit and prevent signal reception. This problem was eliminated by adding flat ribbon fairing to the cable.

As part of initial tests of the Veritas system a set of trials were made in the Oslo fiord with an underwater unit suspended below a vessel. Path lengths varied from 50 to 140M with the underwater unit located in mid water and the bottom. Initial problems with the transducers were overcome by juryrigging an alternate set of transducers and two way telemetry functioned at all positions. It was during these tests that signal skewing was identified as a problem and solutions integrated into the system design.

A summary of data telemetry results is shown in figure 60. It should be observed that none of the tests performed to date have evaluated the software for frequency adaptability. Tests on this have been performed in the laboratory only. Also the range of the tests has been primarily limited by the local environment. The upcoming tests off Italy are configured to give longer path lengths.

As referred to above some tests were performed recently in the laboratory on the frequency adaptive software. The multipath and noise were imitated as in the previously described tests. The receiving system did adapt correctly. This is not however a very effective test of how the system would adapt at sea.

Some results on testing of the alternate decoders have been shown earlier. It is worth commenting on some very early sea tests which were made before this particular program began. The object of that experiment was to examine the feasibility of modifying a standard sonobuoy for data communications. The experiment (figure 61) used a free drifting sonobuoy which was modified to transmit a frequency modulated signal at 30 KHz center frequency. The transducer was suspended below at 100M depth. The receiving hydrophone was suspended below a vessel, drifting at distances from 1 KM to 3 KM away. The received signal was recorded directly on a high speed tape recorder. The hydrophones had compliant suspension to reduce surface induced motion. The weather during the experiment was very calm with only slight surface capillary waves.

The recorded signals have been extensively used to examine various decoder concepts. Data from approximately 2.2 KM distance and at data rates up to 2KBaud have been successfully decoded. Of particular interest is the variability of the bit jitter observed.

8. CONCLUSIONS

The concept of an adaptive acoustic communication system for offshore oil applications has been developed. The system uses four parameters of adaptability, namely frequency, data rate, power and type of decoder. The concept development included evaluation of the requirements by analysis of several typical problems (scenarios), creating the hardware and software to allow experimental measurements and the performance of a series of tests.

These tests have demonstrated that:

1. Adapting the data rate to accomodate environmental problems, particularly bit jitter, has been achieved.

2. Adapting the power level to allow optimization of the energy resources has also been achieved.

3. Adapting the frequency to accomodate varying levels of signal to noise and signal to multipath has been partially achieved. The system has been demonstrated under laboratory conditions but sea trials,to date, have not been sufficient to complete the evaluation.

The ambient noise, particularly in shallow water locations typical of offshore oil operations is very time variable in nature and complicates the measurement of signal to noise ratio. The transient nature of the noise also results in the system spending time in evaluating false starts and therefore reducing the probability of accepting a true message. Additional refinement, in terms of longer term analysis of the noise by the system, must be incorporated. There appears to be several tradeoffs when selecting the method of frequency adaption.

As an example of this, a conversation between transceivers which uses the direct selection of frequency by equating the signal to noise and signal to multipath ratios works when the noise is steady in nature but can fail when the noise is variable in time. If the noise is pulsing or cyclical in nature a better intelligence would be to repeat the message with a time sequence to optimize the probability of success.

In many of the tests one of the transceivers observed a much higher noise than the other. In general the "deck" unit, nearer to the surface, observed more noise. In these cases the use of a directional transducer was quite effective in reducing the noise pickup. The directional transducer also improved the signal to multipath ratio.

The computer hardware was very reliable in use and several conclusions can be reached. The use of CMOS devices gives low power consumption and freedom from the need for voltage regulation. The low power also meant an easier task in reducing internal noise propagation.

The software for adaptability of data rate and power and for general operations involves about 1K words of memory. Addition of frequency adaptability requires an additional 1K of memory. Hence the total memory needs are quite modest. If an 8 bit or 16 bit microprocessor were used this would probably translate into approximately 2K to 4K bytes of memory since the 6100 instruction set is quite compact.

Two versions of the acoustic system which use switch selection of the frequency and software selection of the power and data rate are now in the field and are being evaluated for offshore oil applications. One system is designed for short range (500M) operation with high frequency (64-120 KHz) channels and the other is designed for longer range (5000M) operation with lower frequency (8-28 KHz) channels.

The system, when used with the primary decoder, has an upper limit on data rate of about 3 Kbaud. This is sufficient for many operational requirements. Modifications to the system to allow 10 to 12 KBaud are now being considered. This would allow the transmission of compressed, real time, video information.

Several features of the system have been incorporated into a computer modem design which automatically optimizes the link on installation. This application can be considered a natural extension of the work since a hardwired modem, when operating at high data rates, can experience similar problems with noise and extraneous signals.

9. REFERENCES

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10. Terminology

The following abreviations or terms are used in this report.

A/D - analogue to digital, refers to conversion of data from the first to second form.

AGC - automatic gain control

Baud - equivalent to 1 bit per second.

CMOS - complimentary metal oxide semiconductor, type of construction of electronic active components.

CPU - central processing unit.

DMA - direct memory access.

FIFO - first in,first out data storage unit.

LSI - large scale integration.

NRZ - non return to zero, describes a seriel data sequency where the data levels occur directly without any intermediate levels.

PLL - phase locked loop.

PWM - pulse width modulation, a form of serial digital data where the data value is given by the width or duration at one level relative to the other.

PROM - programmable read only memory.

RAM - random access memory (read/w

S/M - ratio of signal to multipath signal strengths, usually given in dB

S/N - ratio of signal to noise.

uPA - micropascal, a unit of sound level equivalent to a plane wave of rms intensity of -100 dB re 1 dyne/sq.cm.

'bit coherence' refers to the observation of sequential bits which within a certain tolerance are equal in duration.

'bit jitter' refers to the variability of bit durations.

TABLE 1. SEA TRIALS.

	Parameters	Type Test	Results
1.	20M Depth U/W on Bottom Variable Range 8 channel 8-120 KHz	One Way Fixed Freq Fixed Data Rate	Data except where shadowed
2.	As Above 8 channel 13-90 KHz	As Above	As above
3.	As Above	One Way Fixed Freq Variable Data Rate	Variable Data rate Successful
4.	>100 M Depth U/W Unit suspended at 100M	As Above	Range Extended to 1.5 KM and Data To 2.9 KBaud occasion.
5.	>300M Depth U/W at 100M	As Above	Range extended to 3 KM Maximum data rate Limited by bit jitter.
6.	As for 4.	One Way Differing frequencies Receiver In Scan	Very few acquisitions.
7.	As above	As Above. Receiver Indicator Added	Large Variations In Ambient Noise and receiver trapped in scan
8.	As Above	One Way. Reprogrammed scan	Approximately 50% Acquisitions.
9.	As Above	One Way Reprogrammed evaluate	Approximately 75% Acquisitions.

TABLE 1. (Continued) SEA TRIALS.

Parameters	Type Test	Results
10. As Above	Two Way Fixed Freq.	No Data Cable Strumming Problem
11. As Above	Two Way Fixed Frequency Variable Data Rate	50 % Replies On Same Frequency
12. >300M Depth U/W at 200M	As Above Longer ranges	60 - 70 % Replies Range extended to 2.9 KM. Data Rates Limited by jitter
13. 50-140M Depth Variable depth U/W	As Above Vertical Transmission	Data successful at all depths.
14. 1000M Depth U/W at 100M	One Way One Frequency Variable Data Rates Acoustic Signal Recorded	Recordings made Up to 2.2 KM

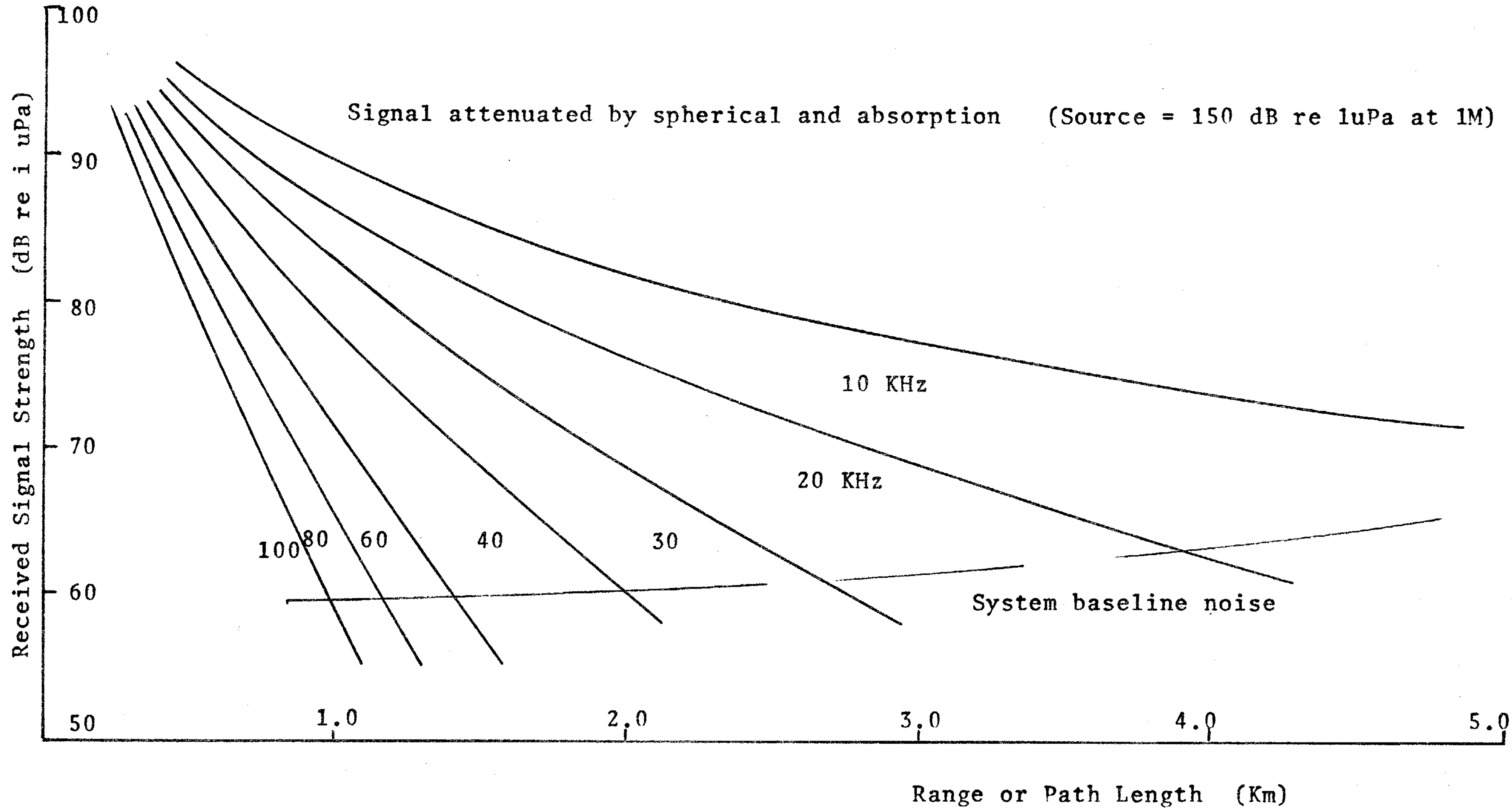


Figure 1. System Scaling From Spreading and Absorption Loss

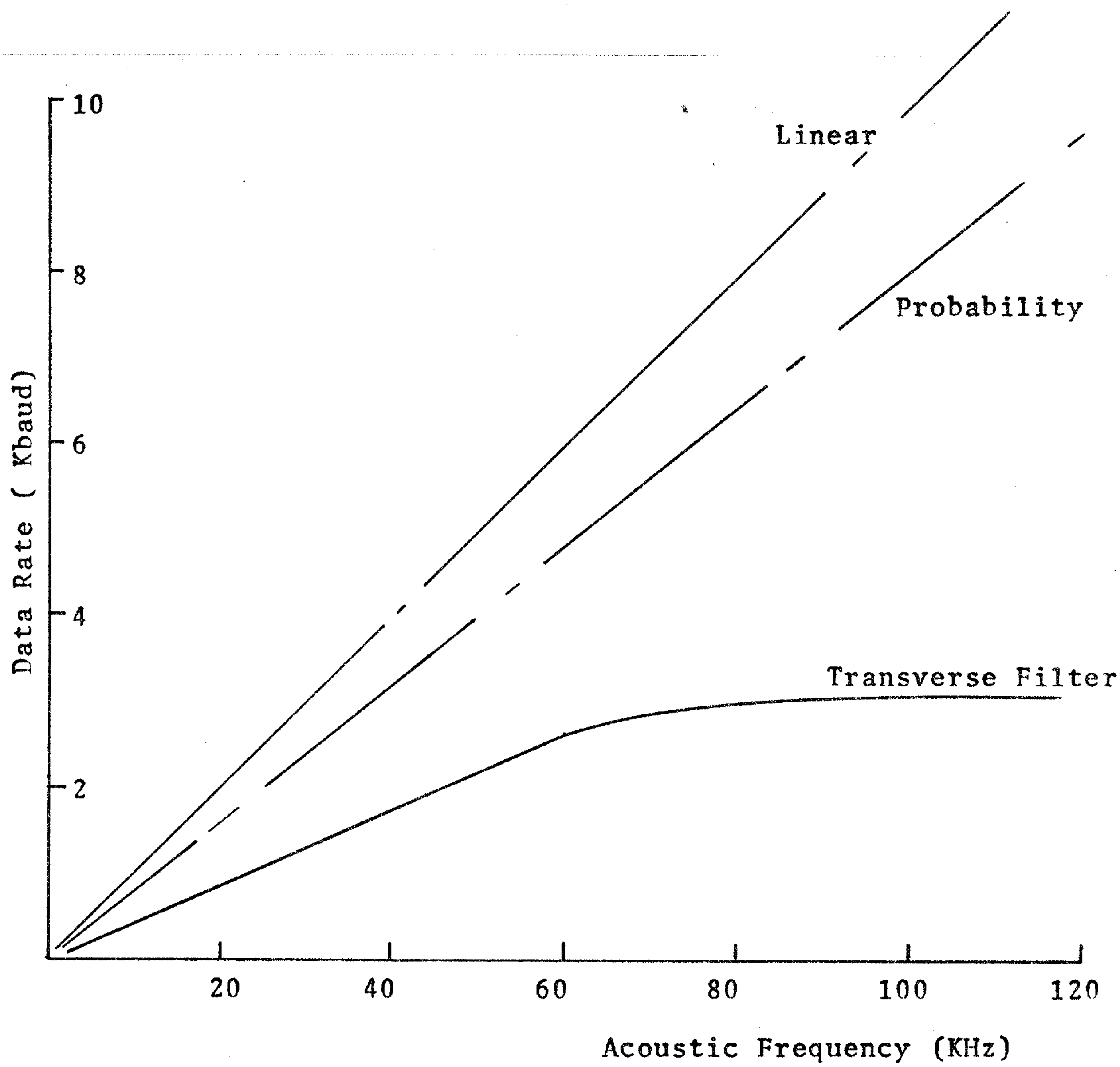


Figure 2. Data Rate Limitations For Different Decoders.

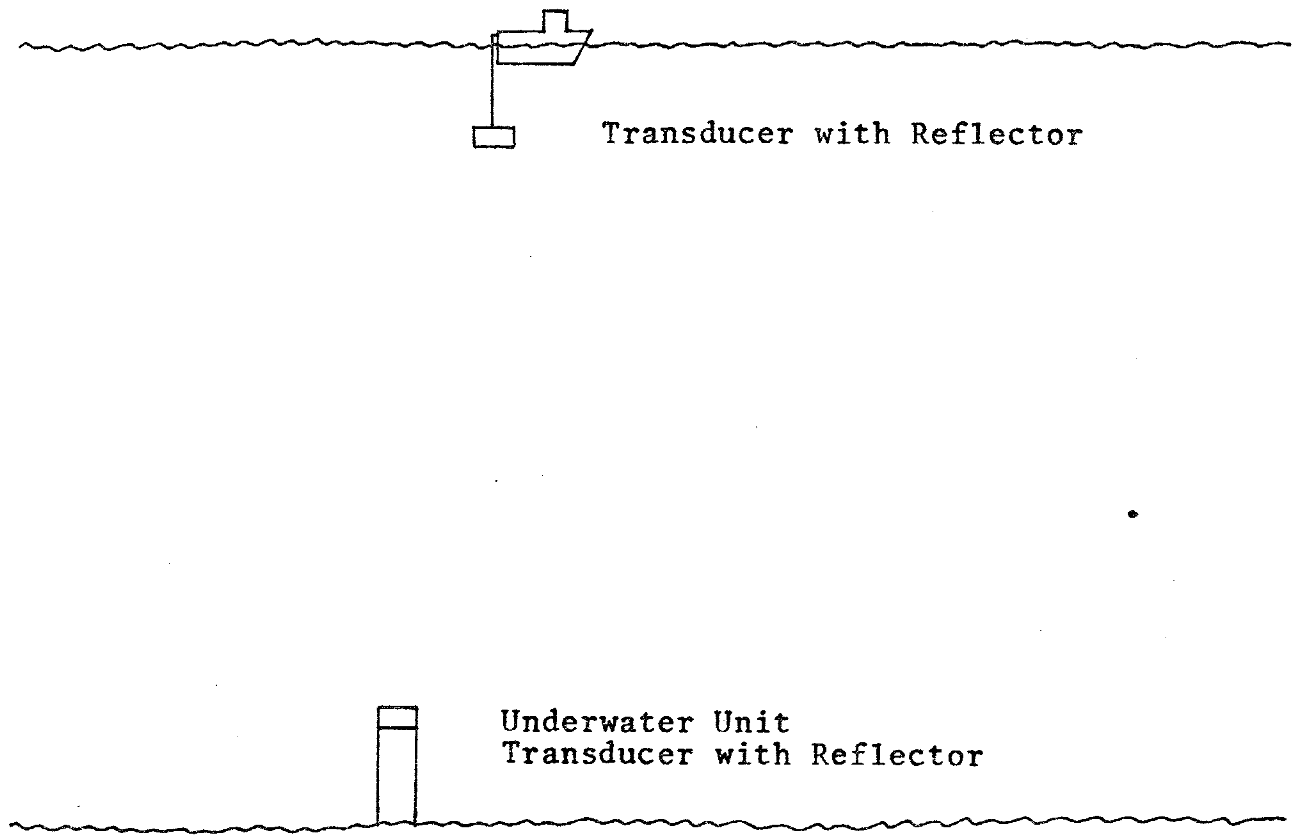


Figure 3. Scenario 1. Ocean Floor to Surface Telemetry

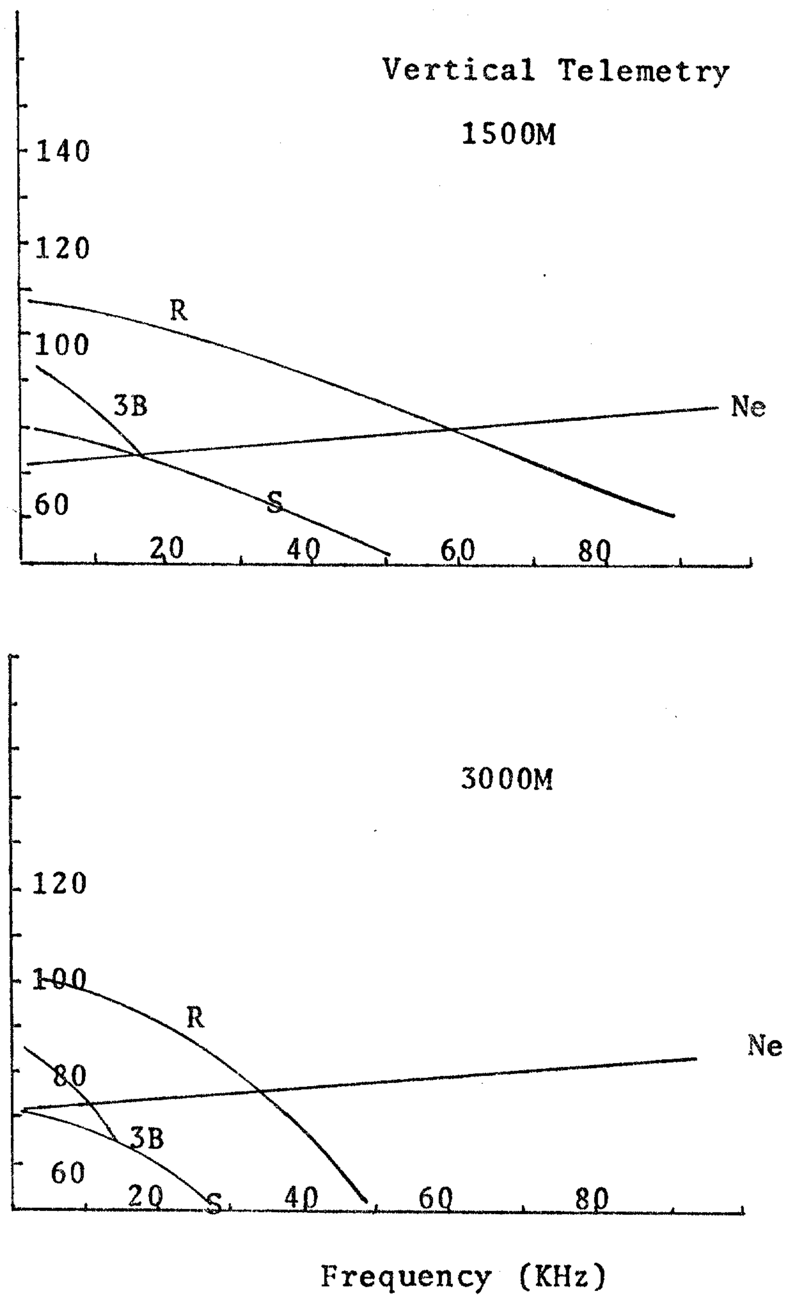


Figure 4.
Acoustic Signals For Vertical Telemetry

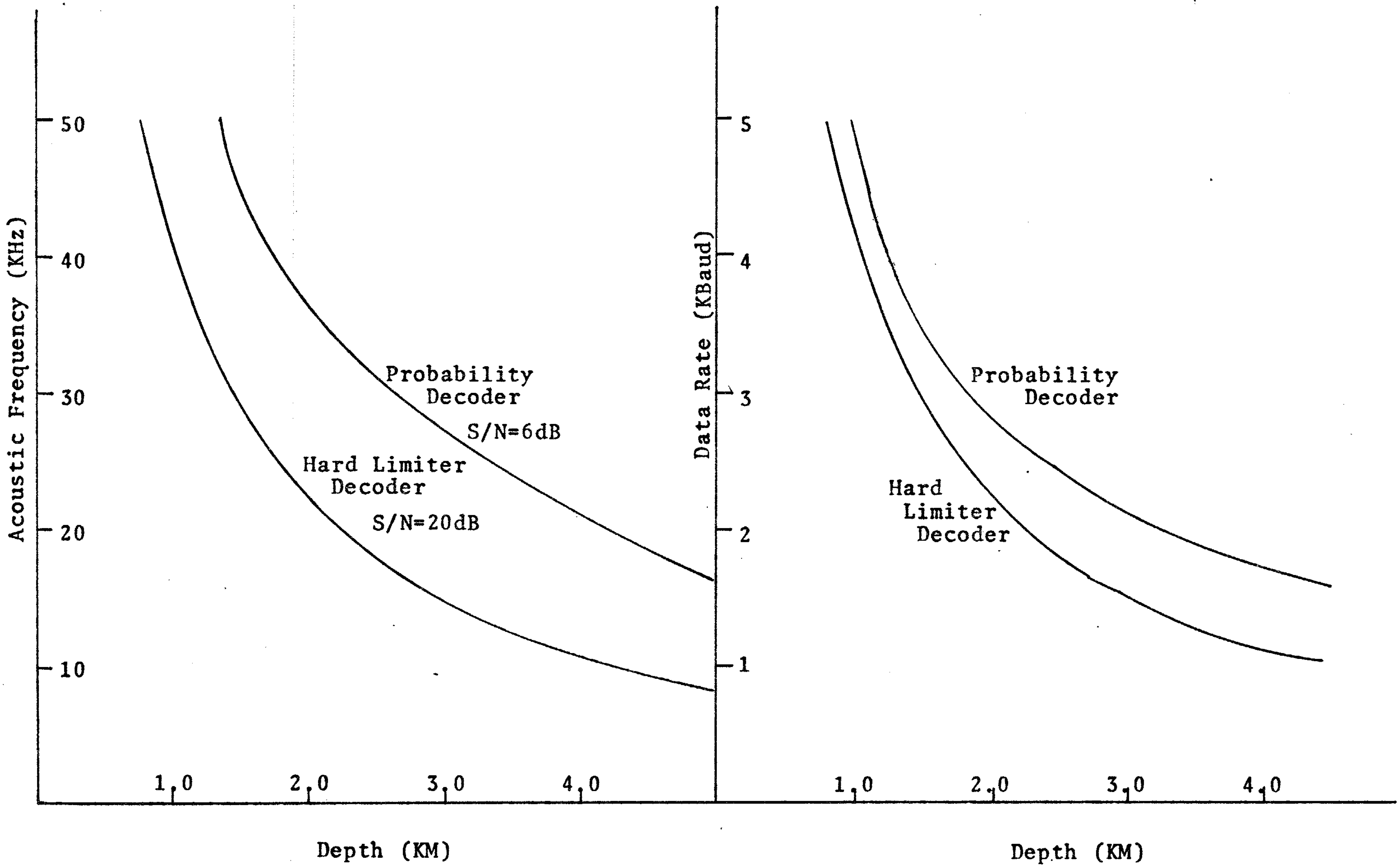


Figure 5. Typical Frequency And Data Rate Limits For Scenario 1.

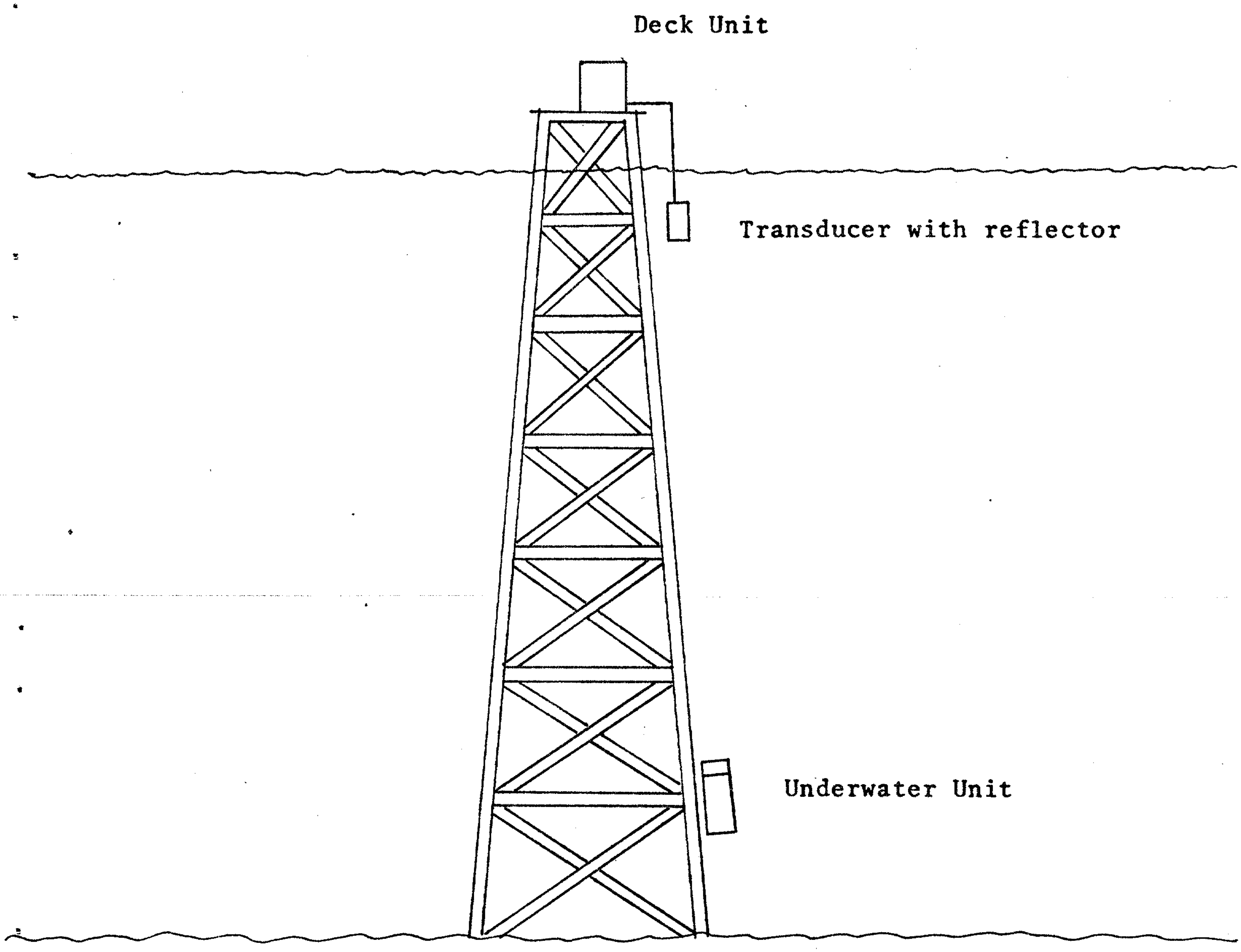


Figure 6. Telemetry On Offshore Platforms.
(Scenario 2.)

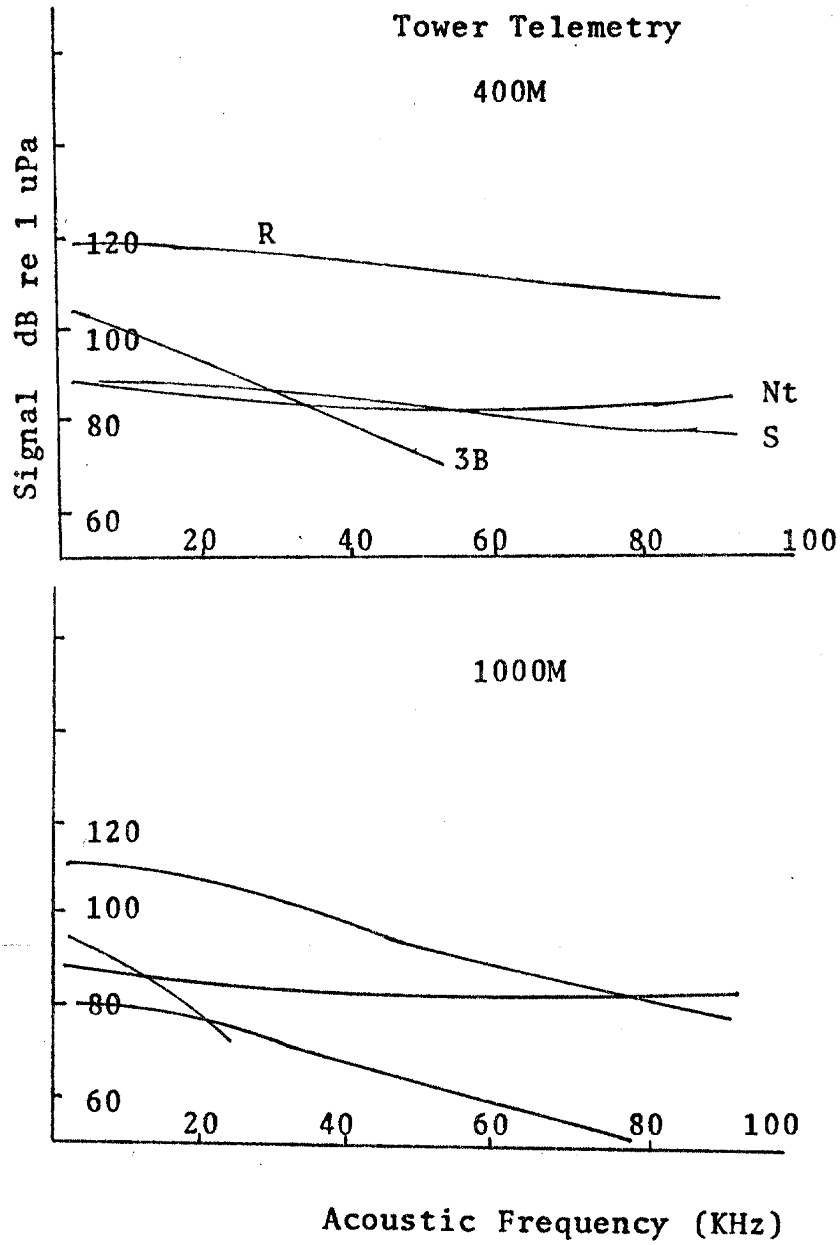


Figure 7. Examples Of Signal Strengths For Scenario 2

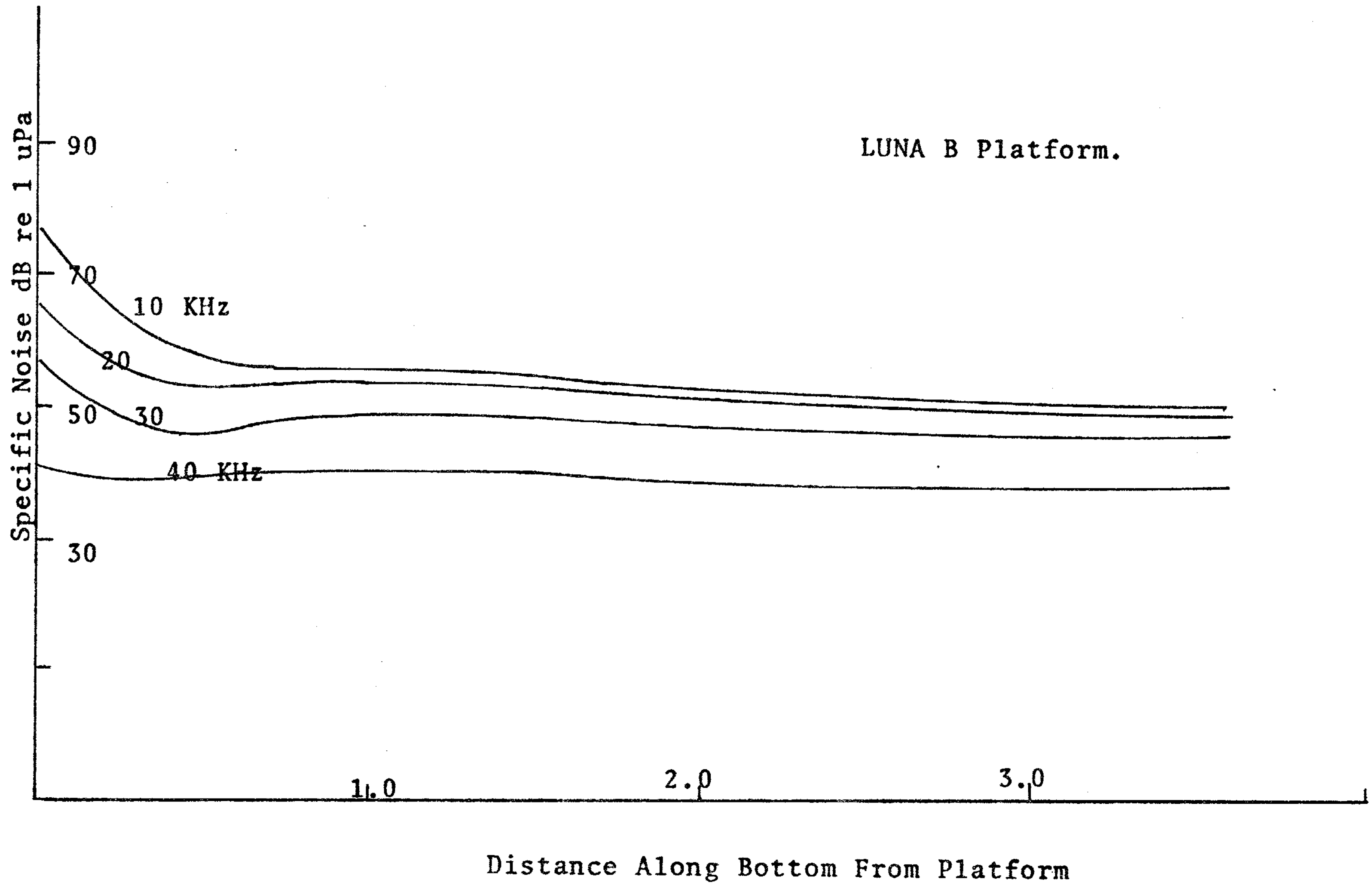


Figure 8. Example Of Ambient Noise In Vicinity Of An Offshore Platform.

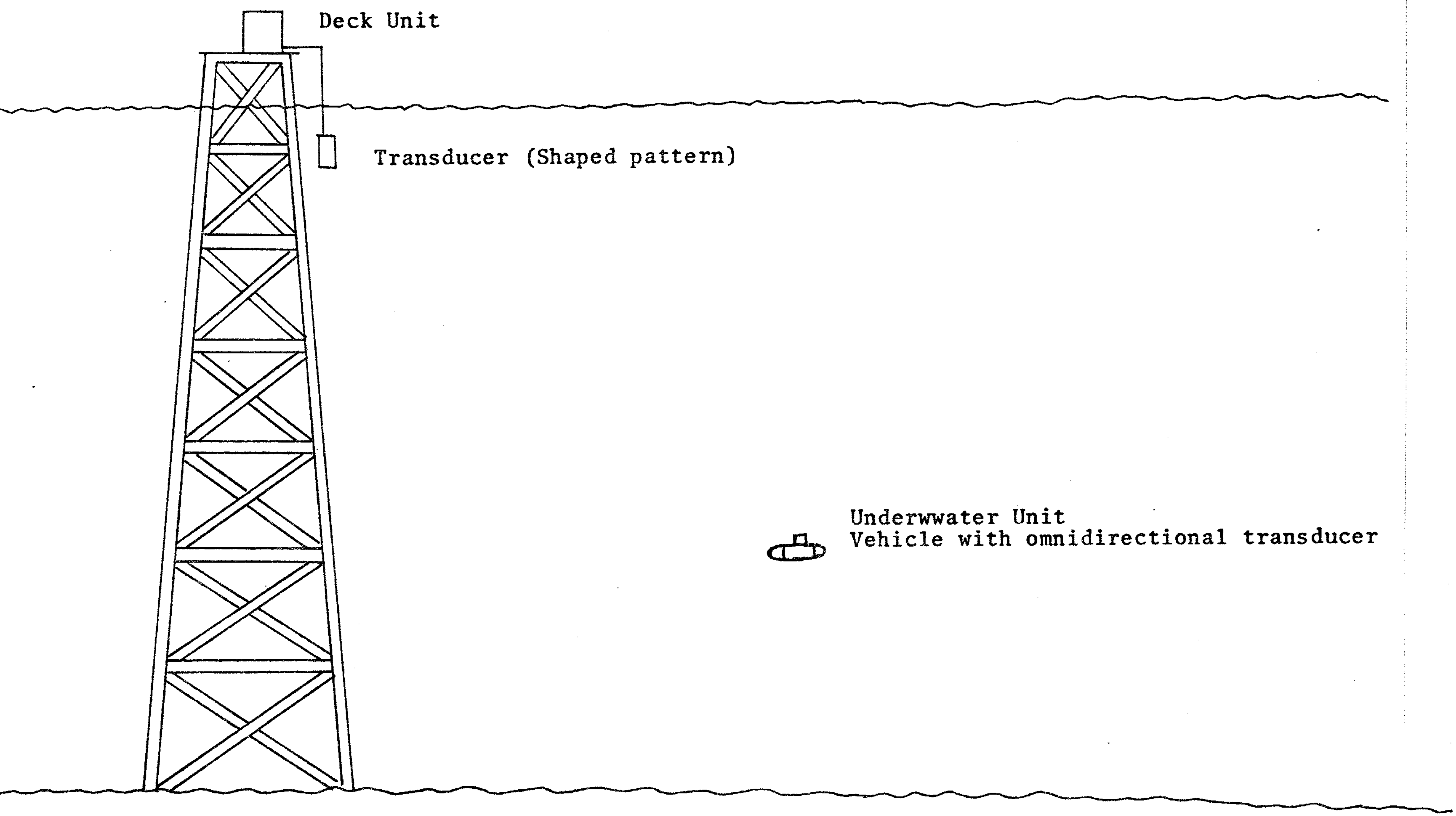


Figure 9. Scenario 3. Platform to Vehicle Communications.

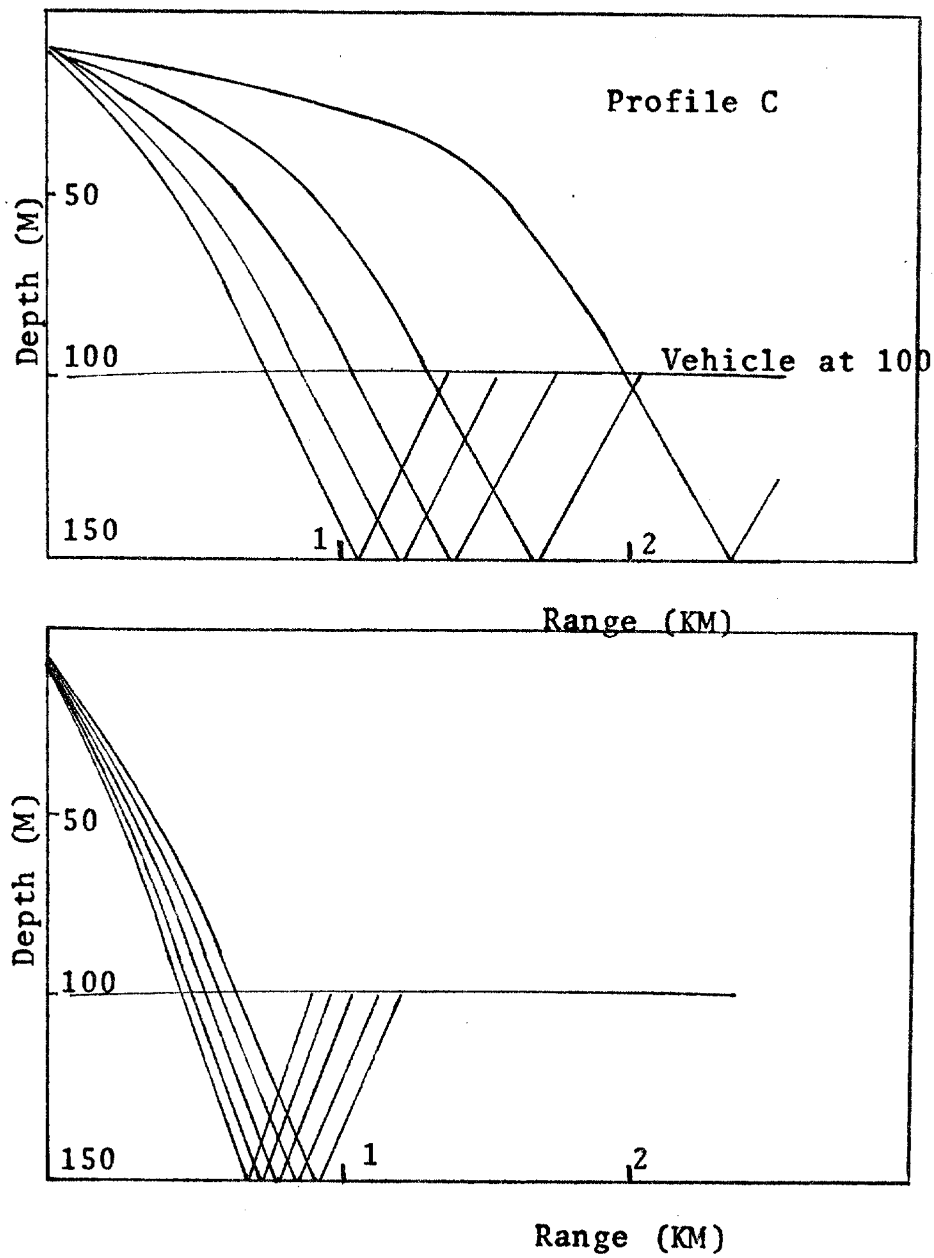


Figure 10. Examples of Ray Paths for Scenario 3.

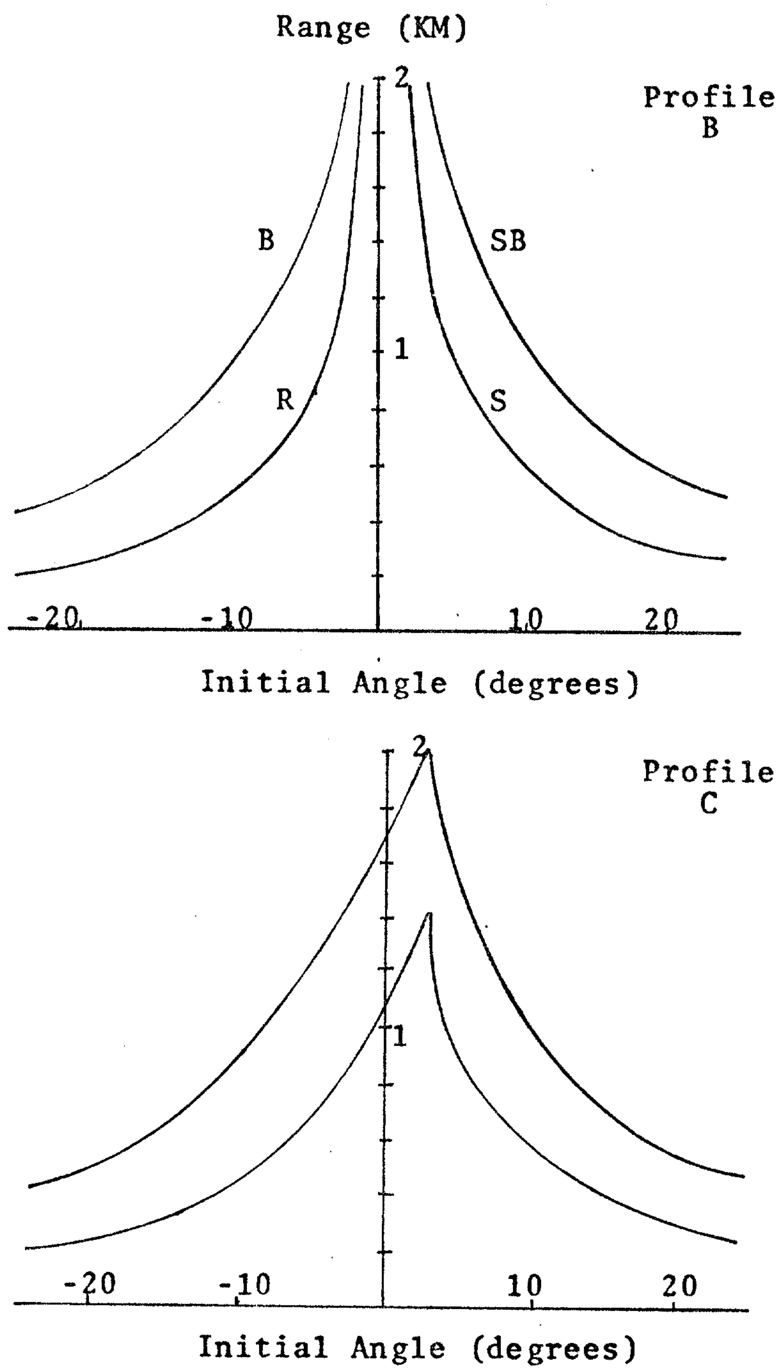
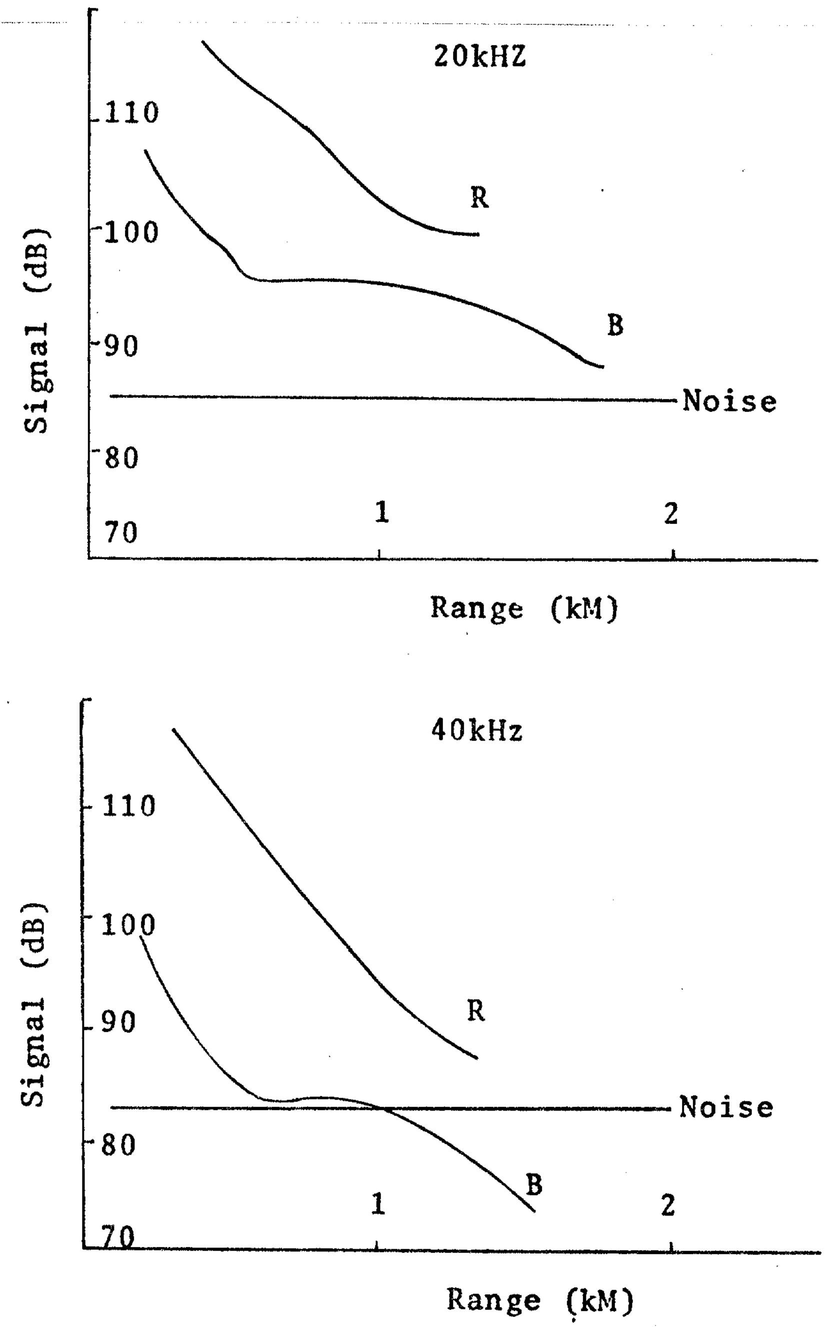


Figure 11. Range Versus Initial Angle.



Water Depth 150M
Vehicle at 100M
Sound Velocity Profile B (Figure 17)
Mud Bottom

Figure 12. Examples of Signal Strength as a Function of Range.

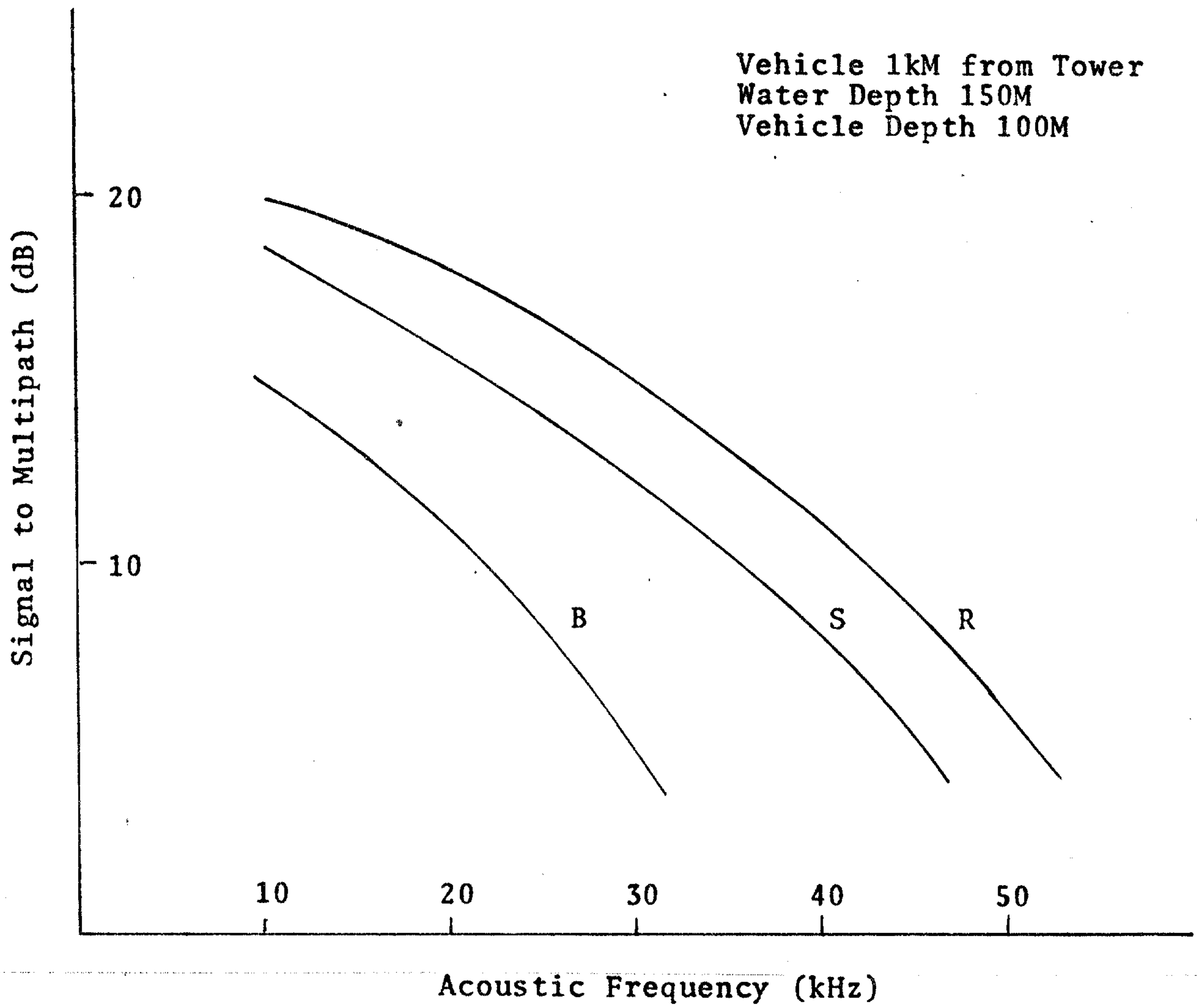


Figure 13. Signal Variations With Frequency at 1kM

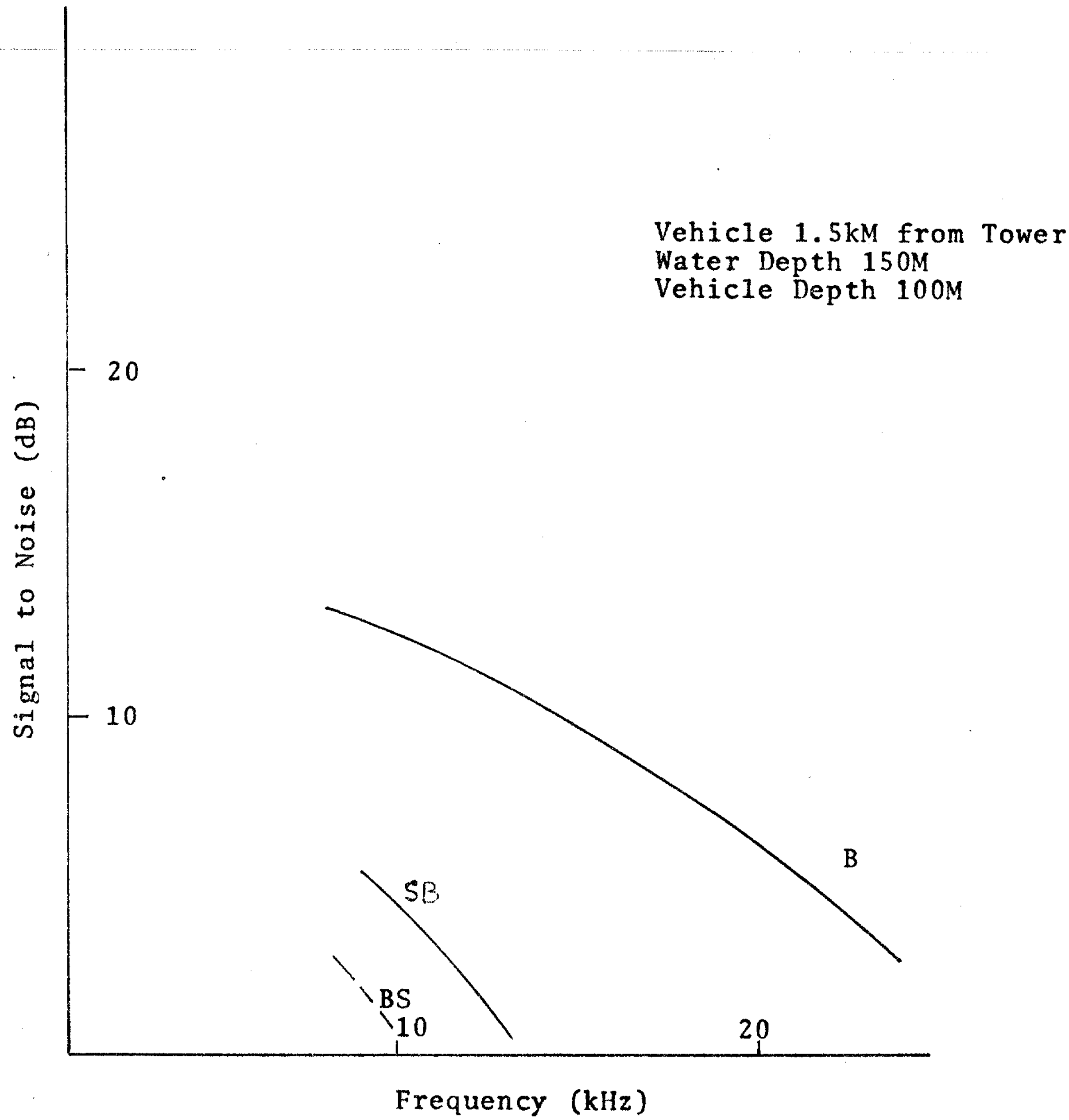
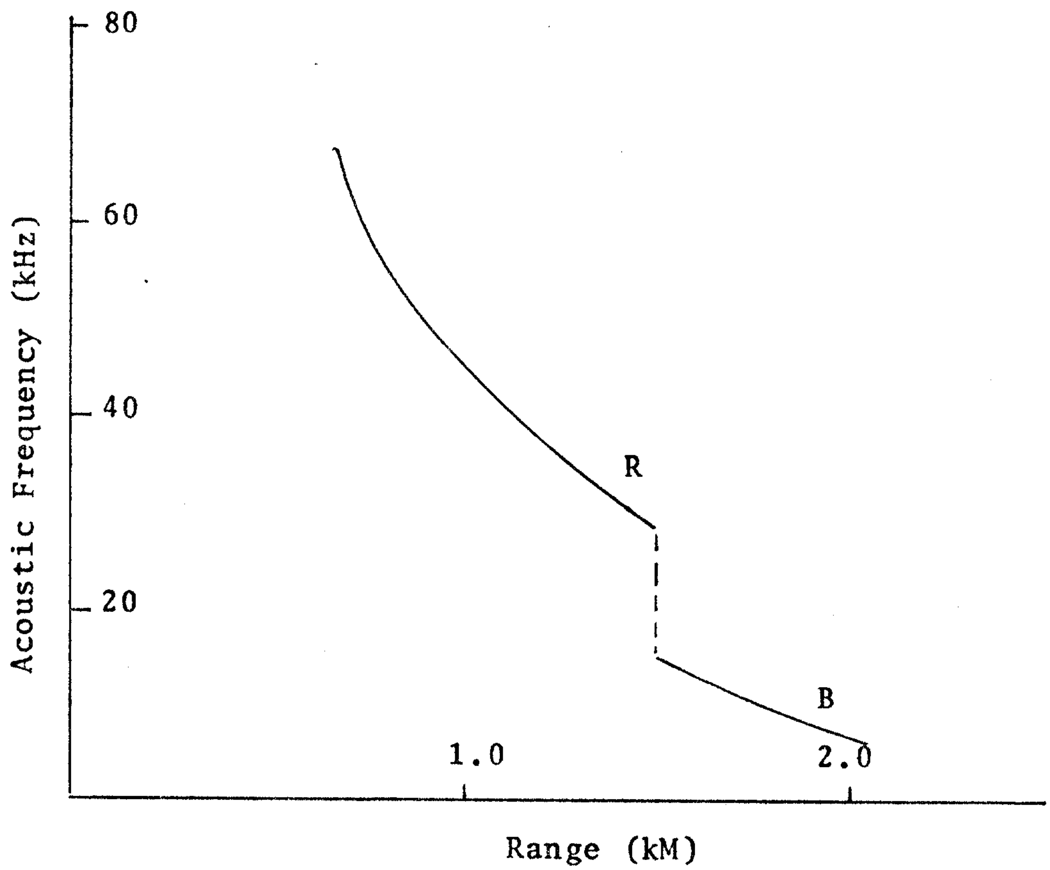


Figure 14. Signal Variations With Frequency at 1.5kM



Vehicle Depth 100M
Water Depth 150M

Figure 15. Effect of Varying Range on Acoustic Frequency Selection

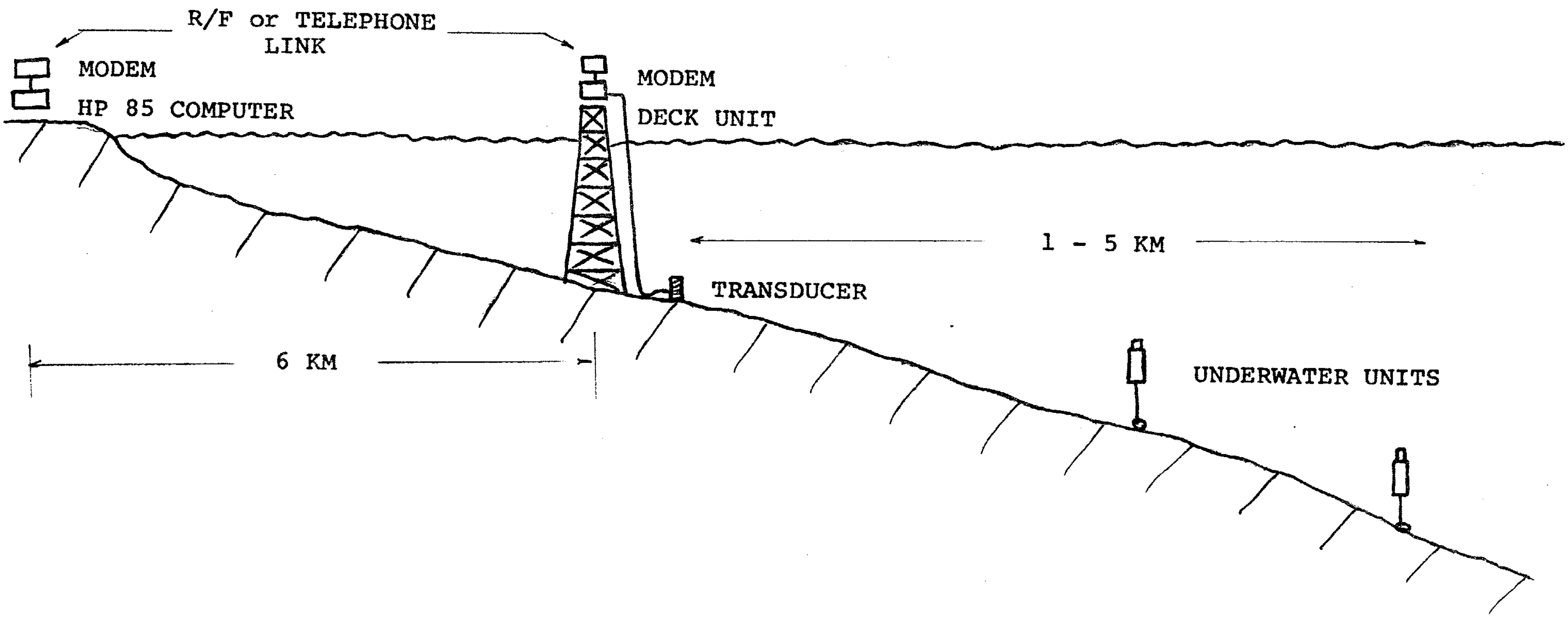


Figure 16. Scenario 4. Platform to Pipeline Communications.

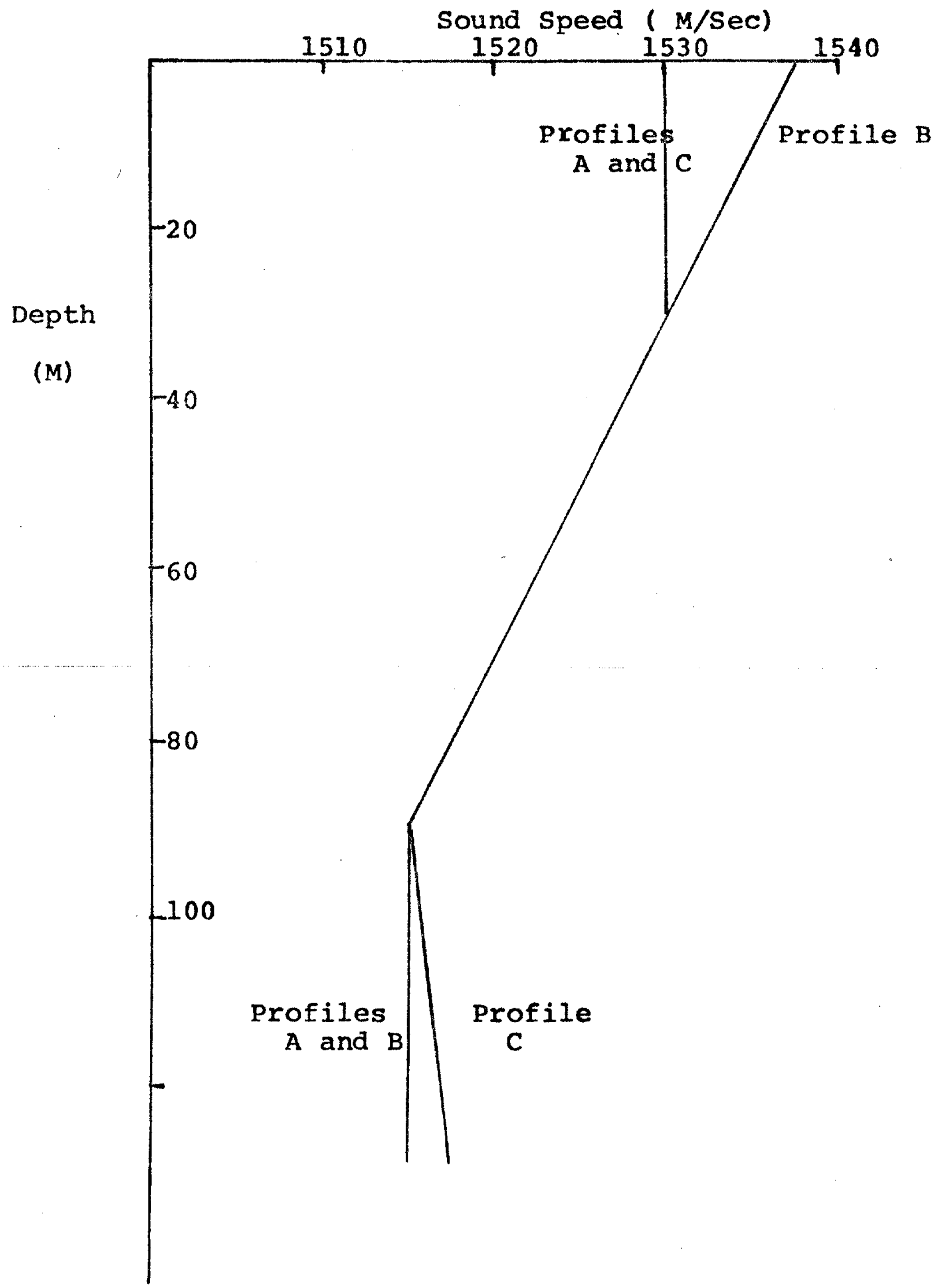


Figure 17. Sound Velocity Profiles for Scenario 4.

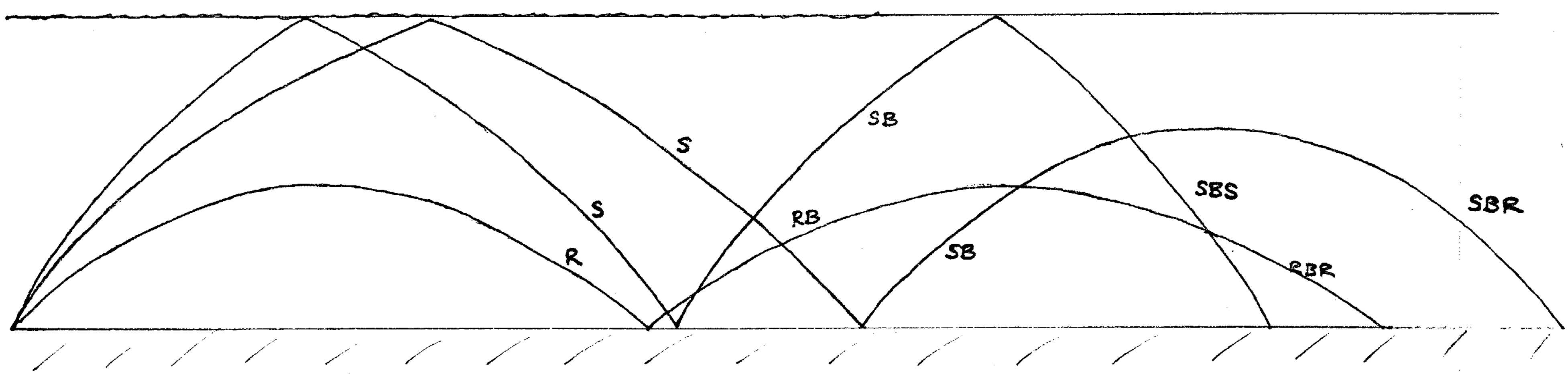


Figure 18. Ray Paths for Scenario 4.

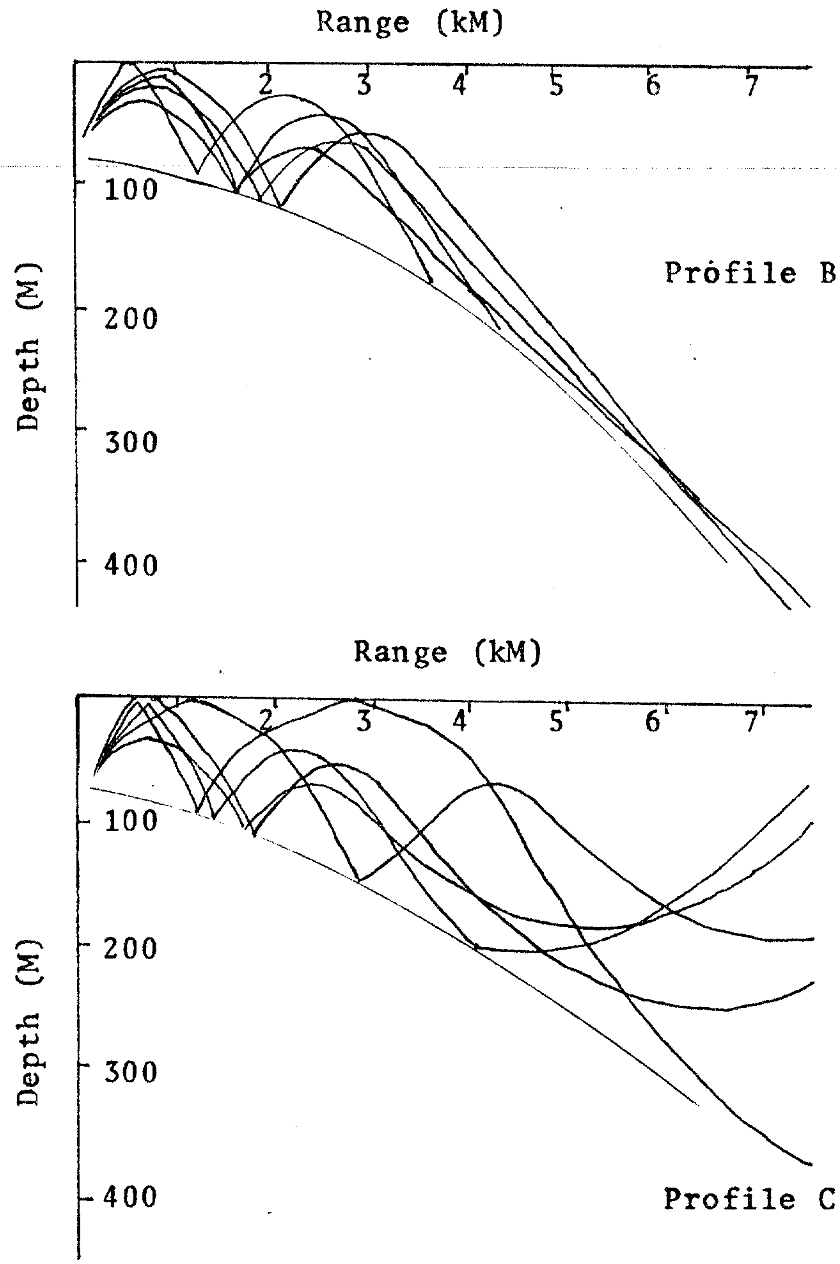


Figure 19. Acoustic Ray Paths for Scenario 4.

Figure 20. Ray Histories for Scenario 4.

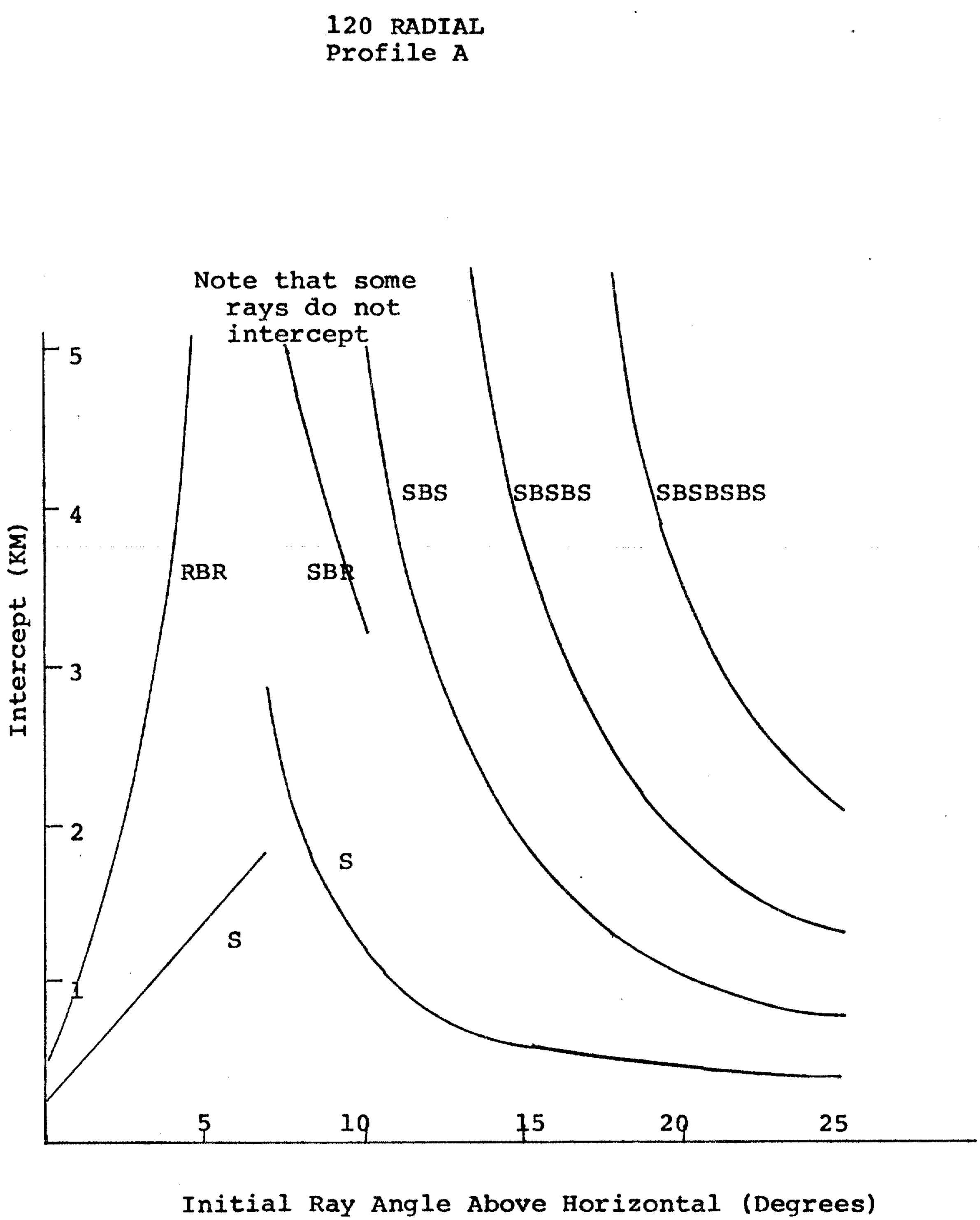


Figure 21. Ray Histories for Scenario 4.

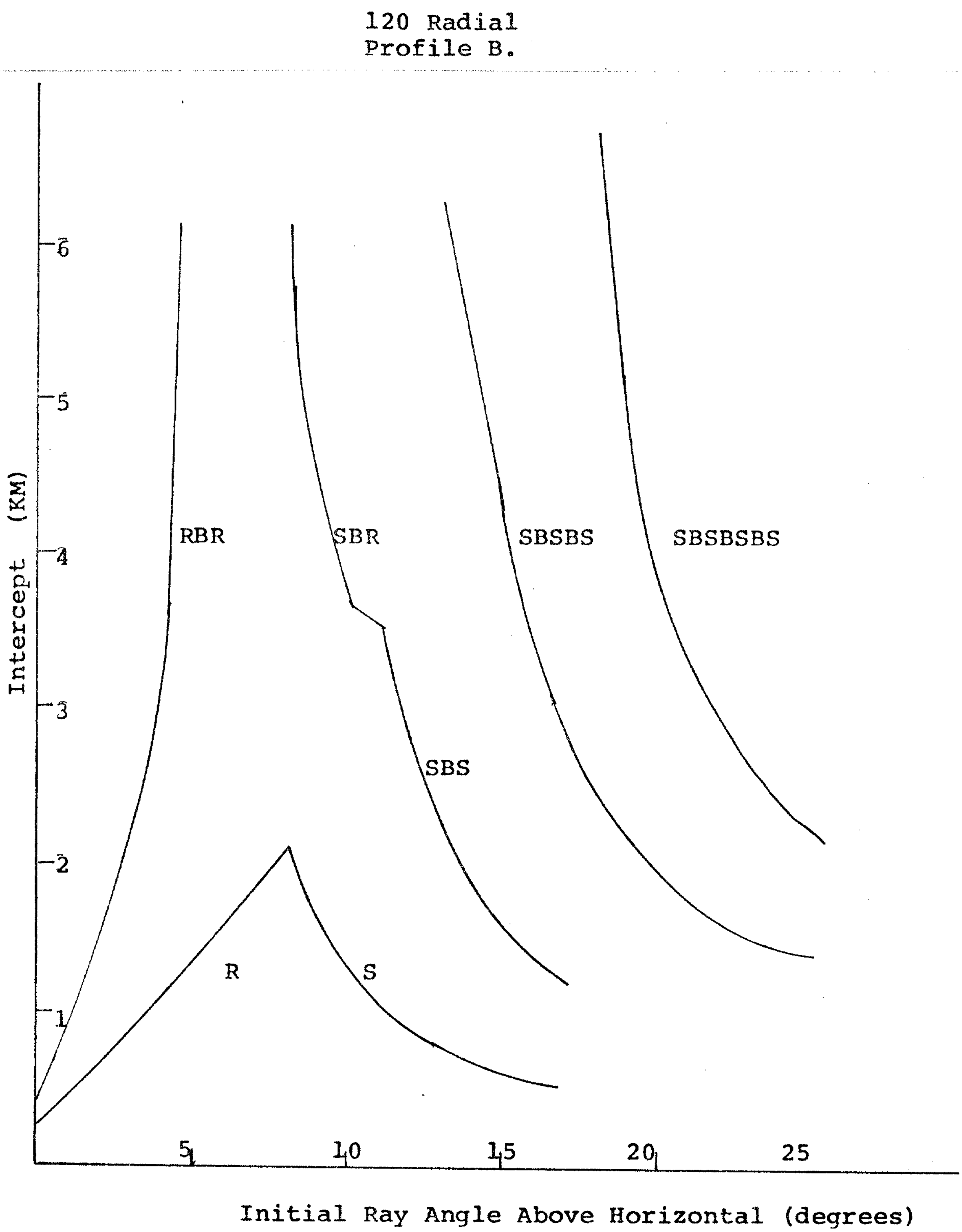
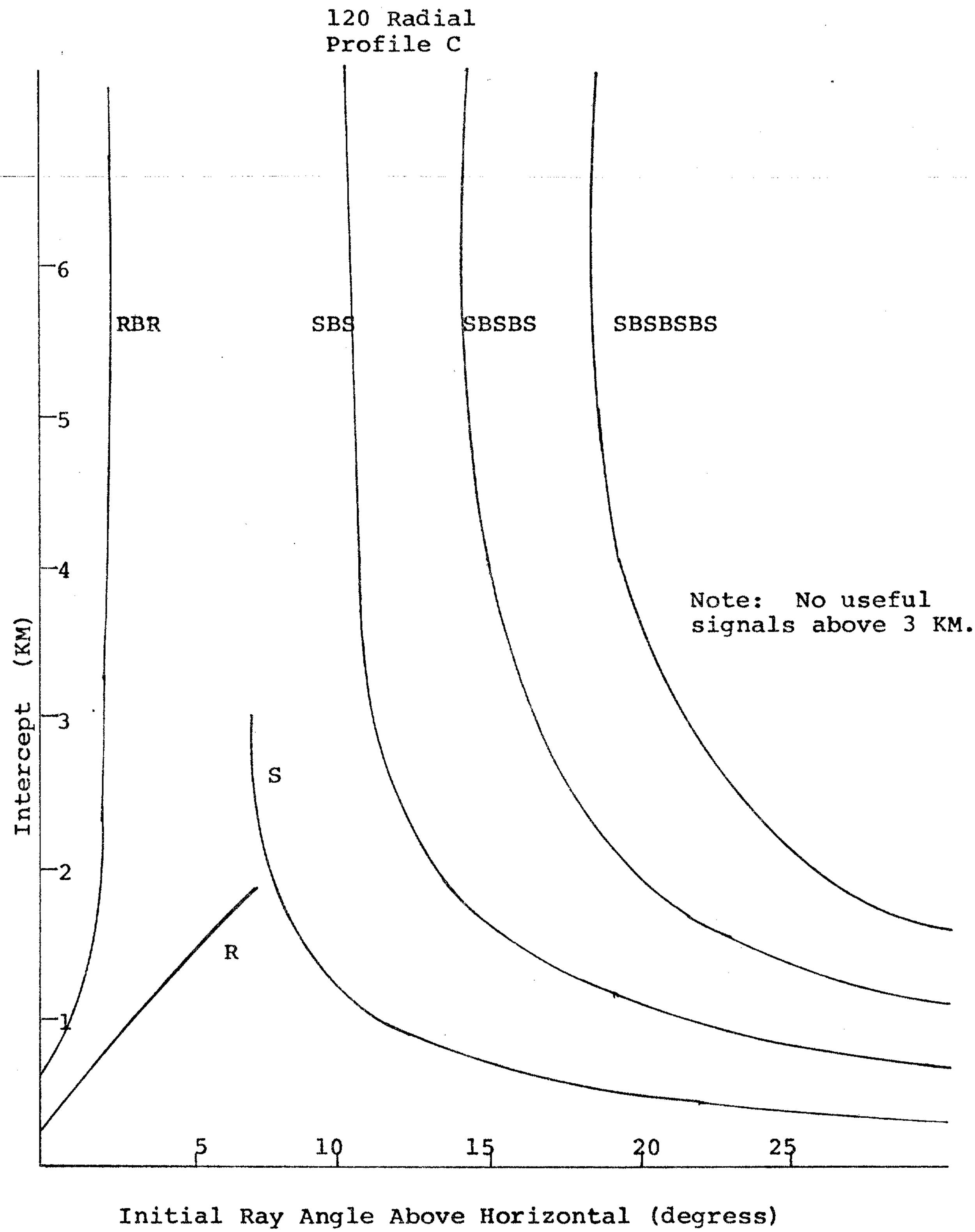


Figure 22. Ray Histories for Scenario 4.



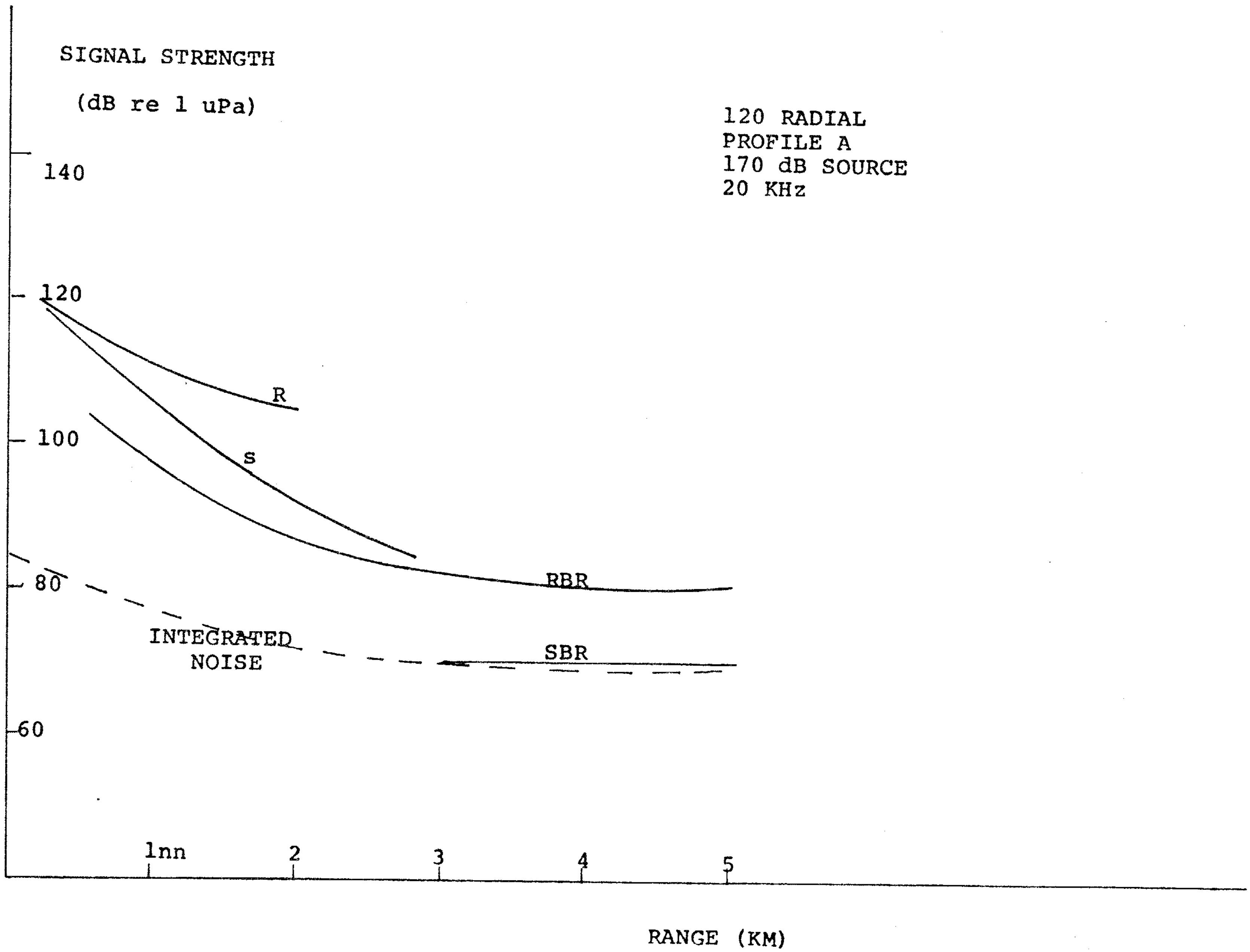


Figure 23. Signal Strengths for Secnario 4.

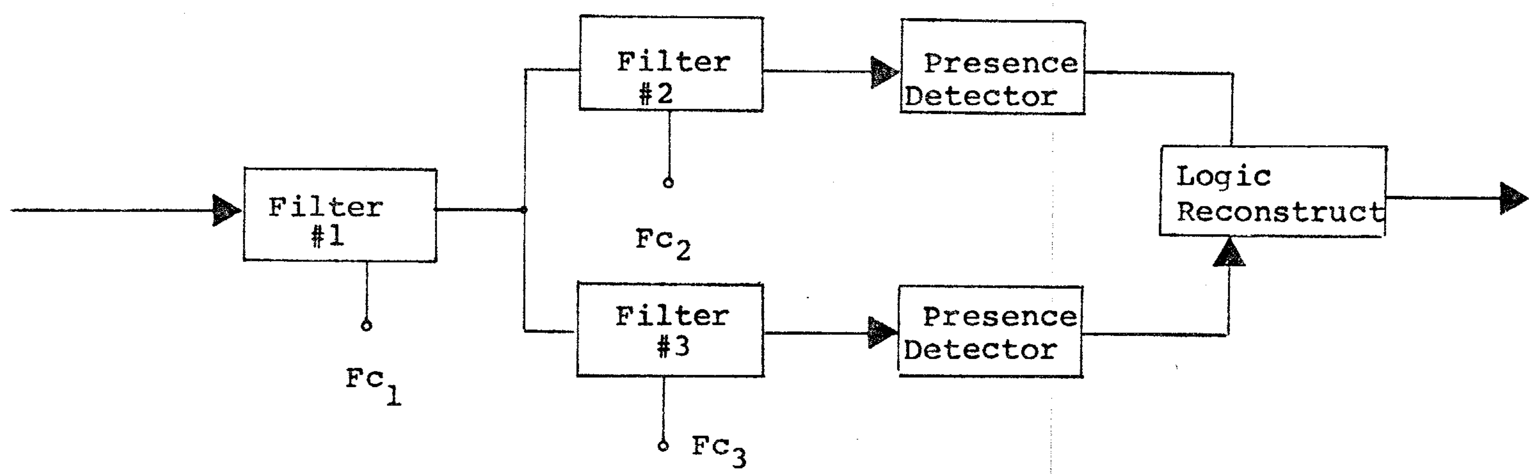


Figure 24. Transverse Filter Arrangement.

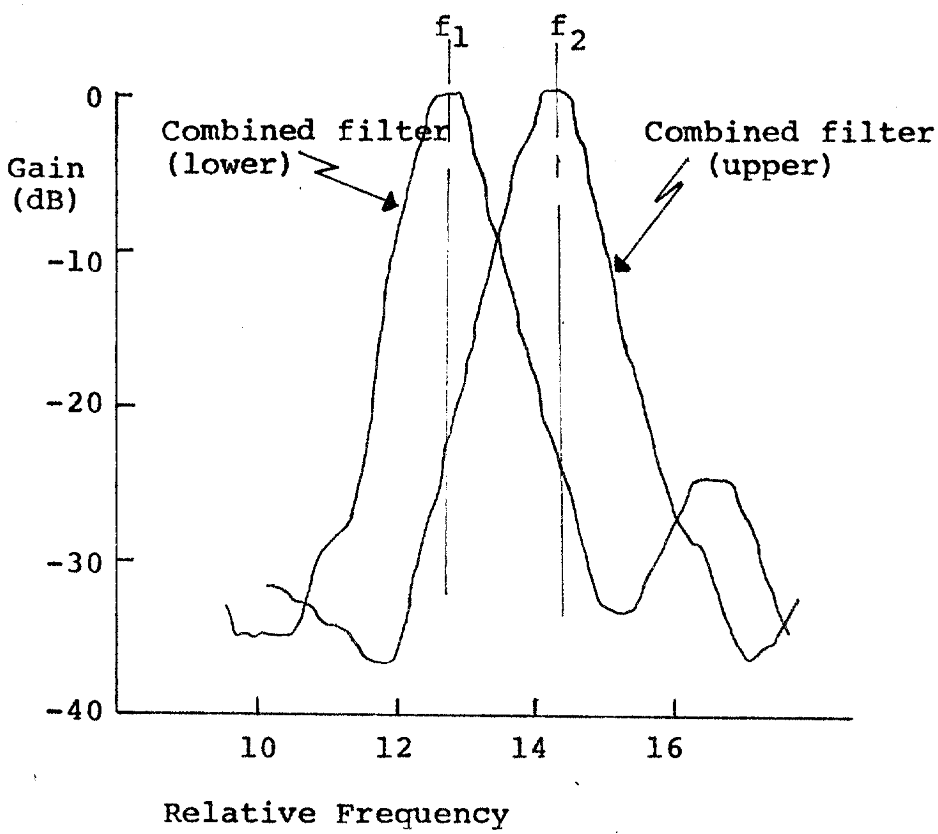


Figure 25. Cascaded Transverse Filter Response.

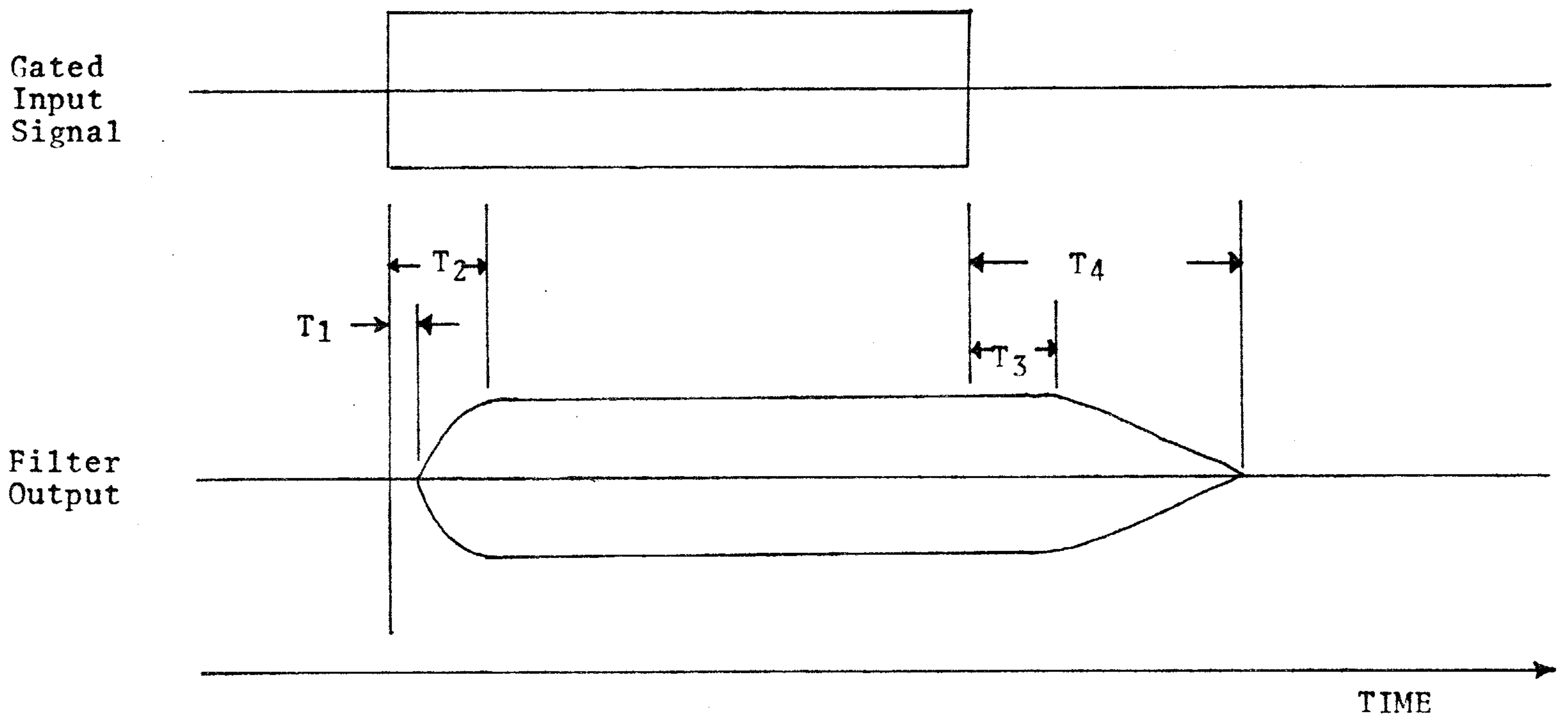


Figure 26. Transient Filter Response.

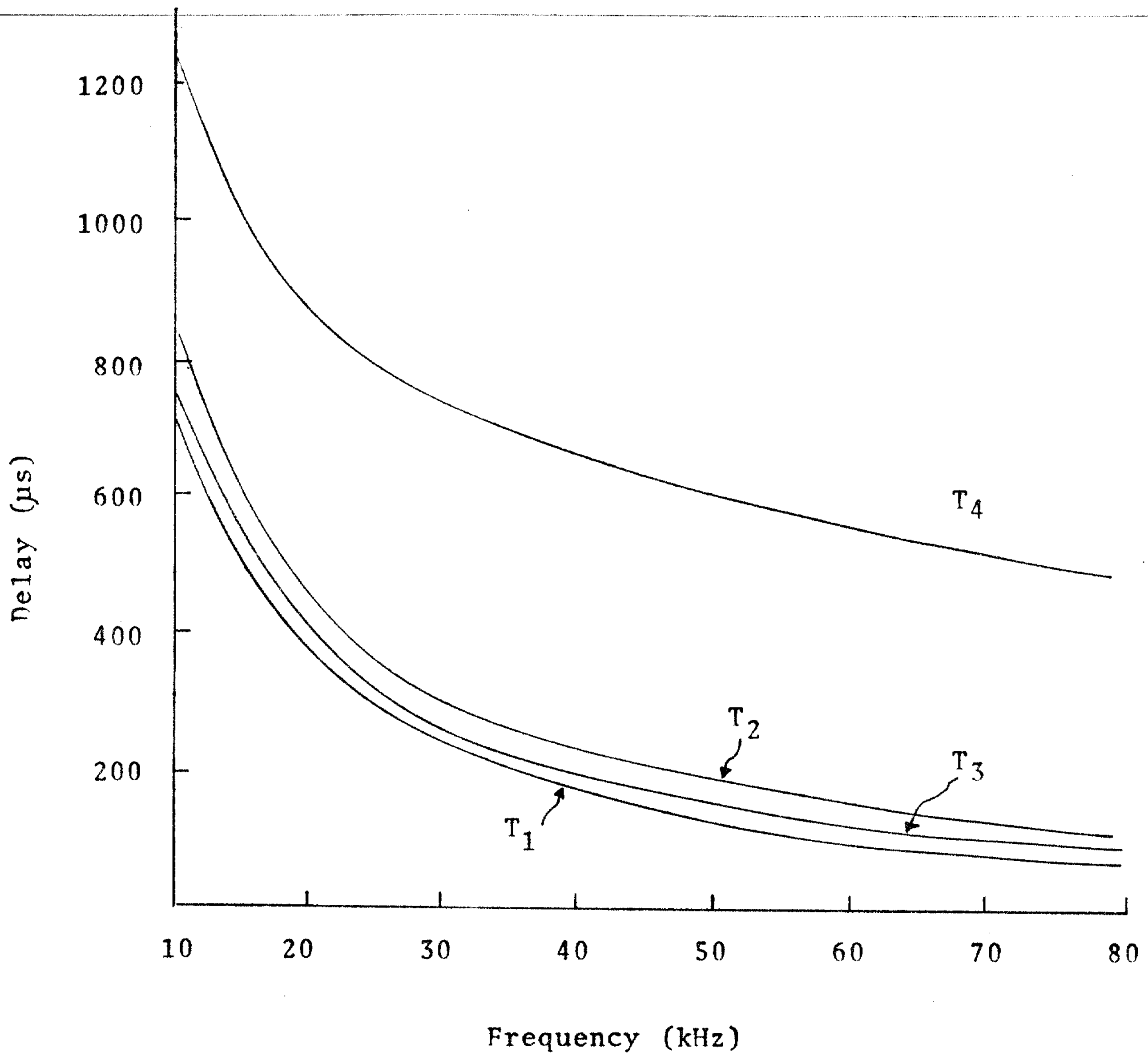


Figure 27. Reaction Times for Filter Array

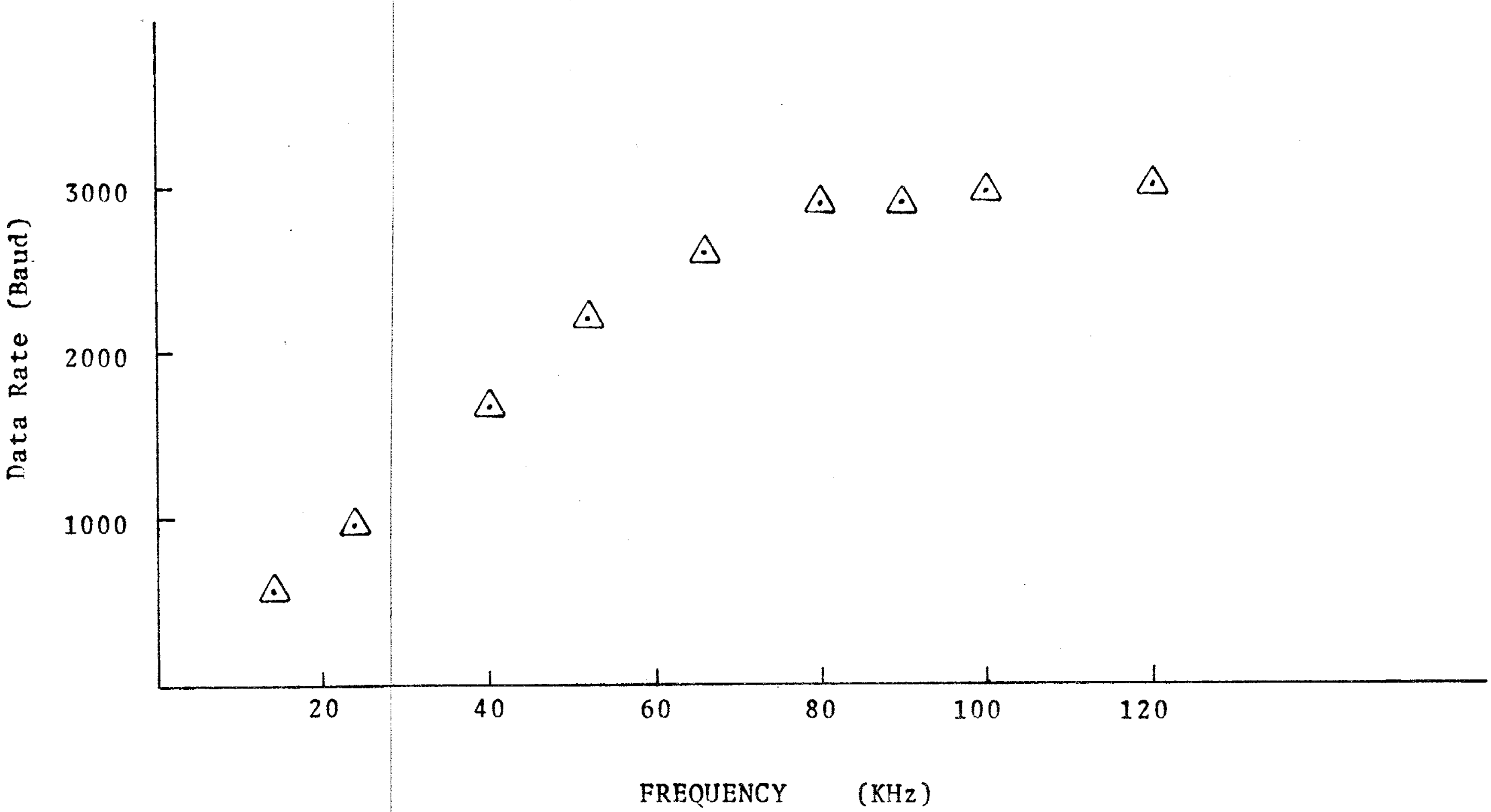


Figure 28. Measured Maximum Data Rate Through Filter Array

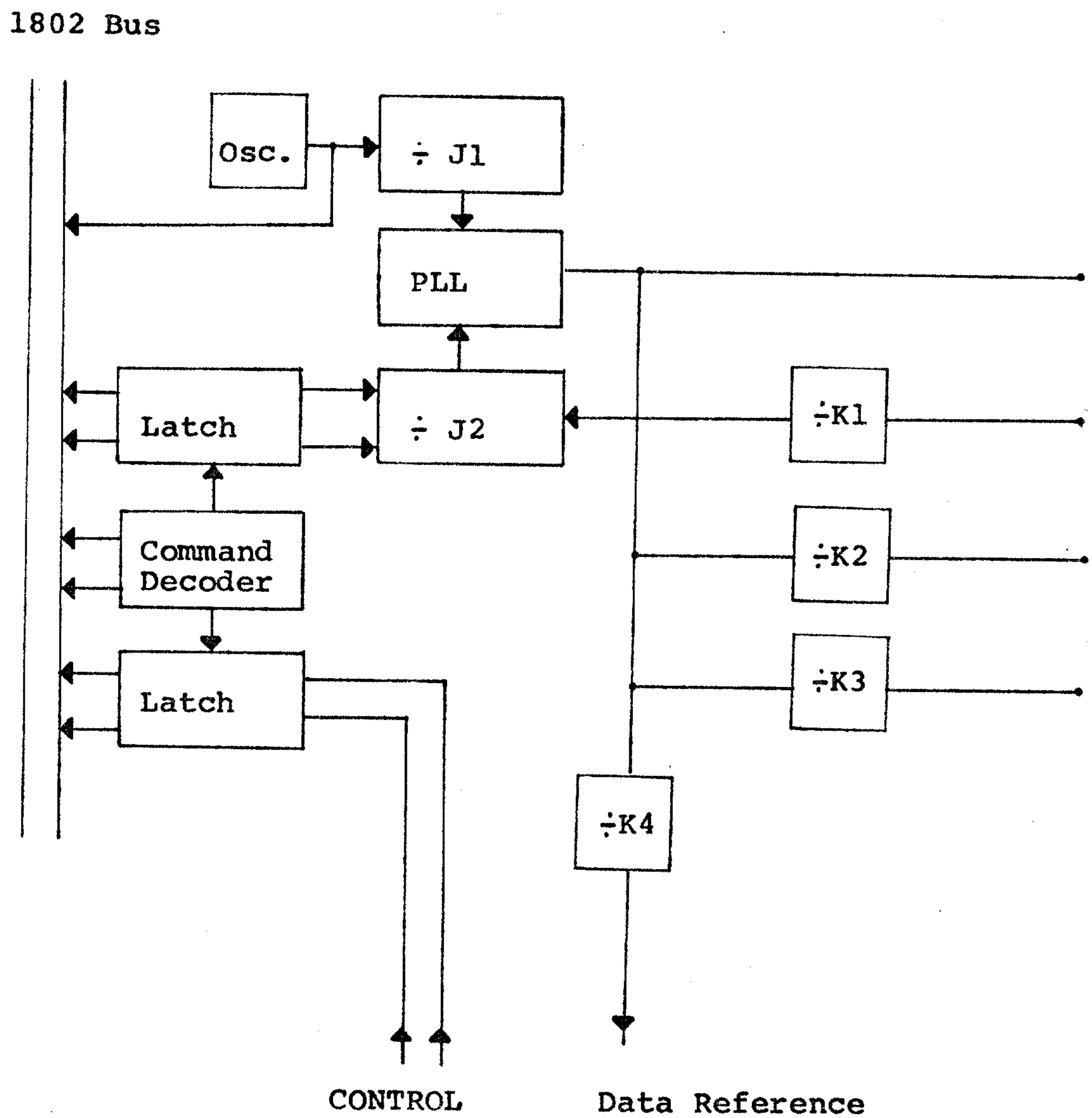


Figure 29. Frequency Synthesizer.

J	FX	ACOUSTIC FREQUENCIES FC	FL	FH	CLOCK FREQUENCIES FC1	FC2	FC3
10	156.25	13.02	12.02	14.20	250.00	208.33	178.57
11	171.88	14.32	13.22	15.63	275.00	229.17	196.43
12	187.50	15.63	14.42	17.05	300.00	250.00	214.29
13	203.13	16.93	15.63	18.47	325.00	270.83	232.14
14	218.75	18.23	16.83	19.89	350.00	291.67	250.00
15	234.38	19.53	18.03	21.31	375.00	312.50	267.86
16	250.00	20.83	19.23	22.73	400.00	333.33	285.71
17	265.63	22.14	20.43	24.15	425.00	354.17	303.57
18	281.25	23.44	21.63	25.57	450.00	375.00	321.43
19	296.88	24.74	22.84	26.99	475.00	395.83	339.29
20	312.50	26.04	24.04	28.41	500.00	416.67	357.14
21	328.13	27.34	25.24	29.83	525.00	437.50	375.00
22	343.75	28.65	26.44	31.25	550.00	458.33	392.86
23	359.38	29.95	27.64	32.67	575.00	479.17	410.71
24	375.00	31.25	28.85	34.09	600.00	500.00	428.57
25	390.63	32.55	30.05	35.51	625.00	520.83	446.43
26	406.25	33.85	31.25	36.93	650.00	541.67	464.29
27	421.88	35.16	32.45	38.35	675.00	562.50	482.14
28	437.50	36.46	33.65	39.77	700.00	583.33	500.00
29	453.13	37.76	34.86	41.19	725.00	604.17	517.86
30	468.75	39.06	36.06	42.61	750.00	625.00	535.71
31	484.38	40.36	37.26	44.03	775.00	645.83	553.57
32	500.00	41.67	38.46	45.45	800.00	666.67	571.43
33	515.63	42.97	39.66	46.88	825.00	687.50	589.29
34	531.25	44.27	40.87	48.30	850.00	708.33	607.14
35	546.88	45.57	42.07	49.72	875.00	729.17	625.00
36	562.50	46.88	43.27	51.14	900.00	750.00	642.86
37	578.13	48.18	44.47	52.56	925.00	770.83	660.71
38	593.75	49.48	45.67	53.98	950.00	791.67	678.57
39	609.38	50.78	46.88	55.40	975.00	812.50	696.43
40	625.00	52.08	48.08	56.82	1000.00	833.33	714.29
41	640.63	53.39	49.28	58.24	1025.00	854.17	732.14
42	656.25	54.69	50.48	59.66	1050.00	875.00	750.00
43	671.88	55.99	51.68	61.08	1075.00	895.83	767.86
44	687.50	57.29	52.88	62.50	1100.00	916.67	785.71
45	703.13	58.59	54.09	63.92	1125.00	937.50	803.57
46	718.75	59.90	55.29	65.34	1150.00	958.33	821.43
47	734.38	61.20	56.49	66.76	1175.00	979.17	839.29
48	750.00	62.50	57.69	68.18	1200.00	1000.0	857.14
49	765.63	63.80	58.89	69.60	1225.00	1020.8	875.00
50	781.25	65.10	60.10	71.02	1250.00	1041.7	892.86

*NOTE:
1. Reference frequency equals 1MHz.
2. Predivider equals 64.
3. Output is in kHz.

Figure 30. Calculated Output Frequencies from Synthesizer

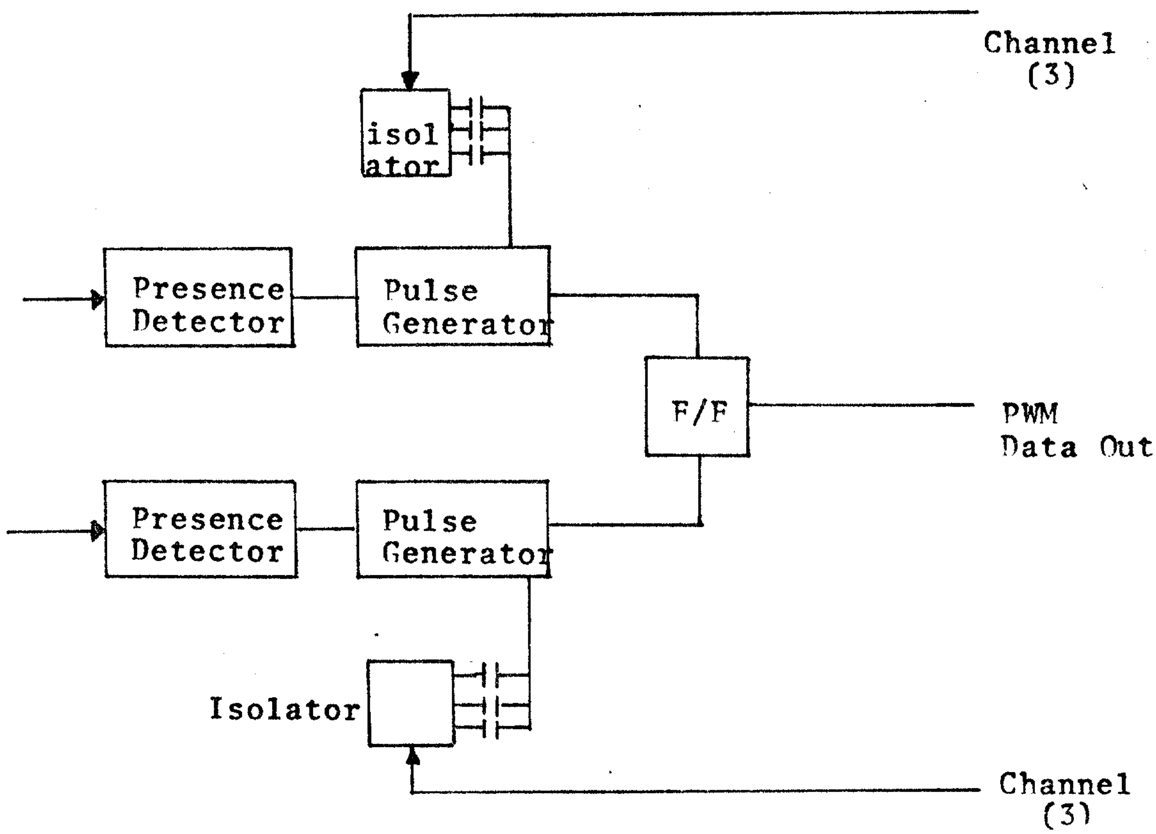


Figure 31. Channel Control Of Decoder

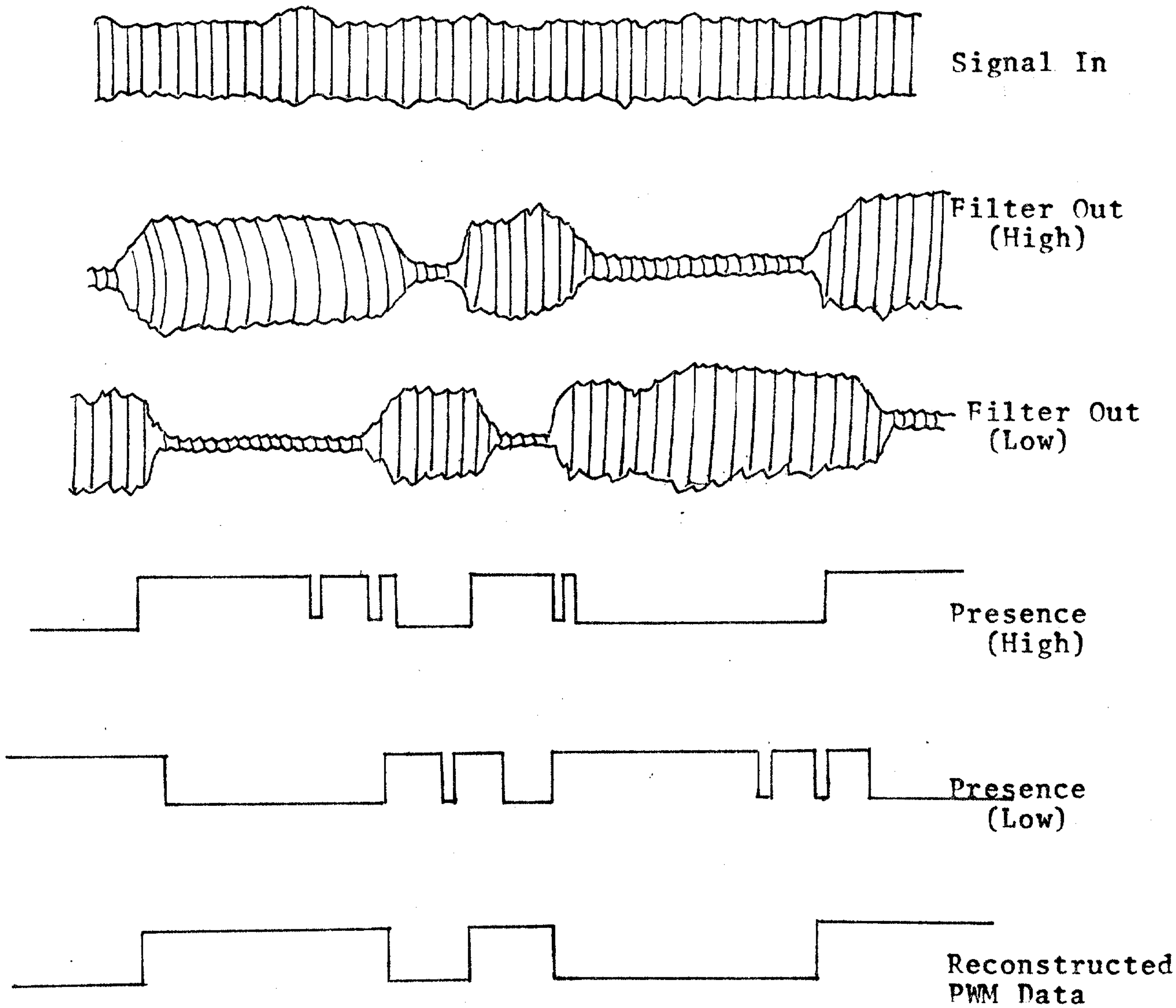
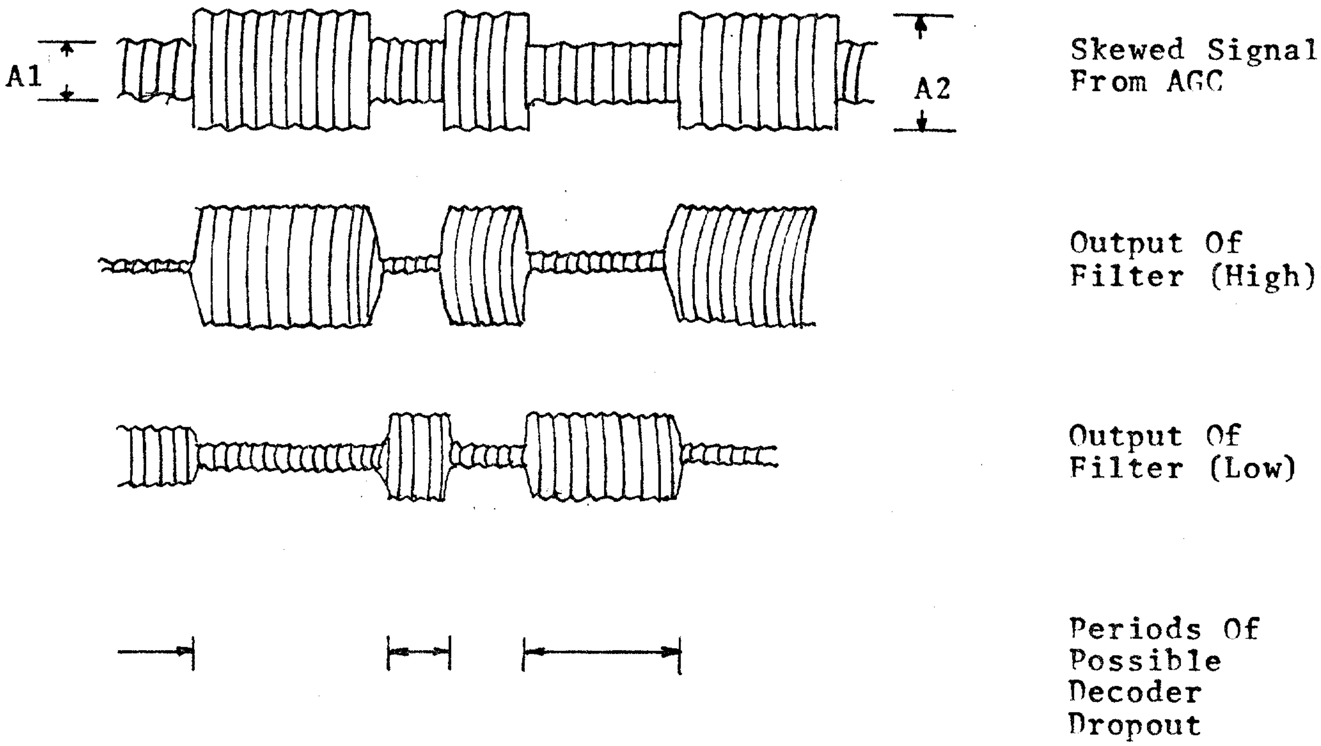
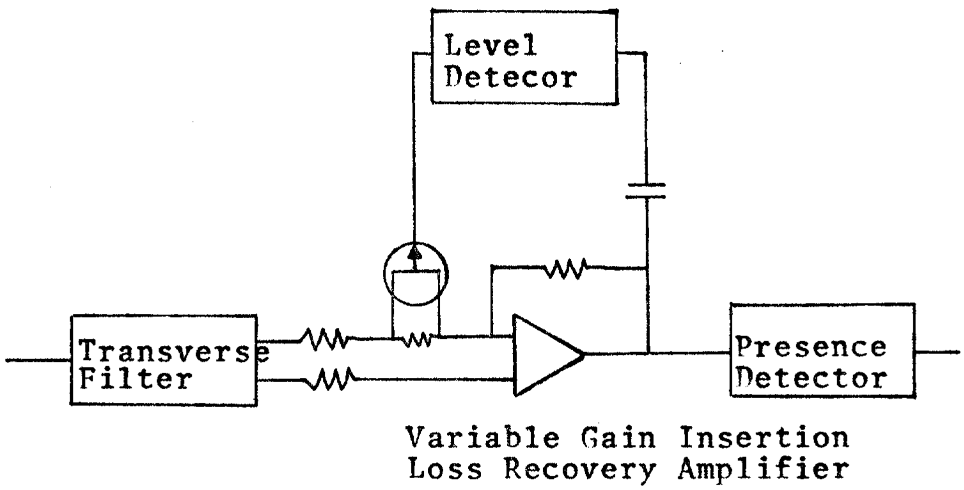


Figure 32. Signal Within The Decoder.



Skew Defined As 20 Log(A2/A1)

Figure 33. Skewed Signal Through Decoder.



Note: Similar Circuitry For Other Signal

Figure 34. Deskewing Technique Within Decoder.

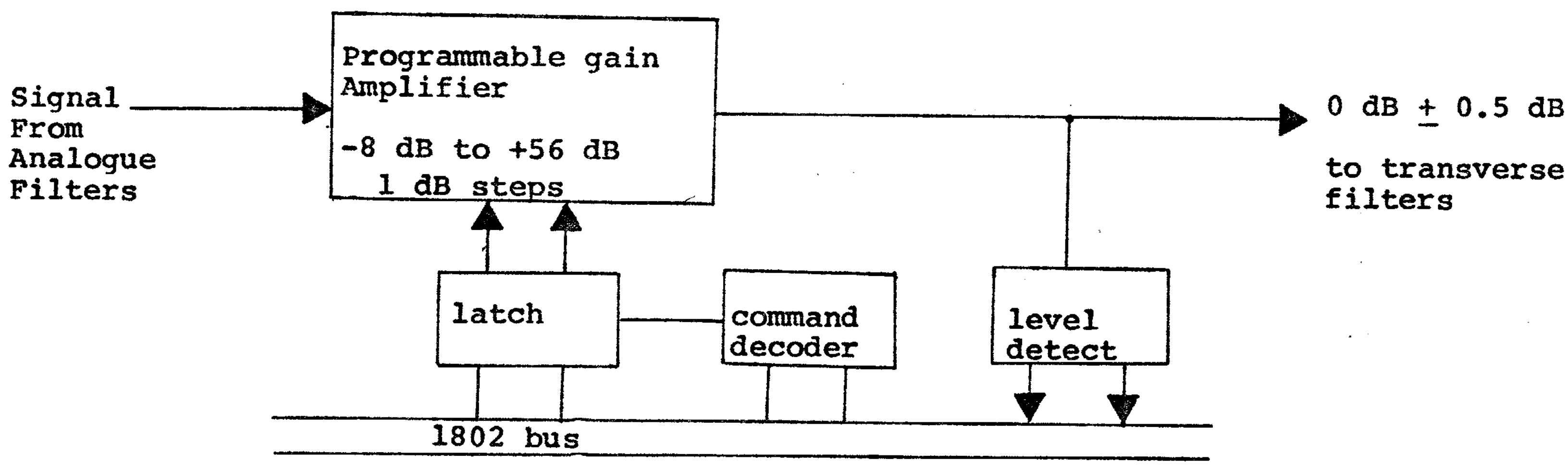


Figure 35. Automatic Gain Control

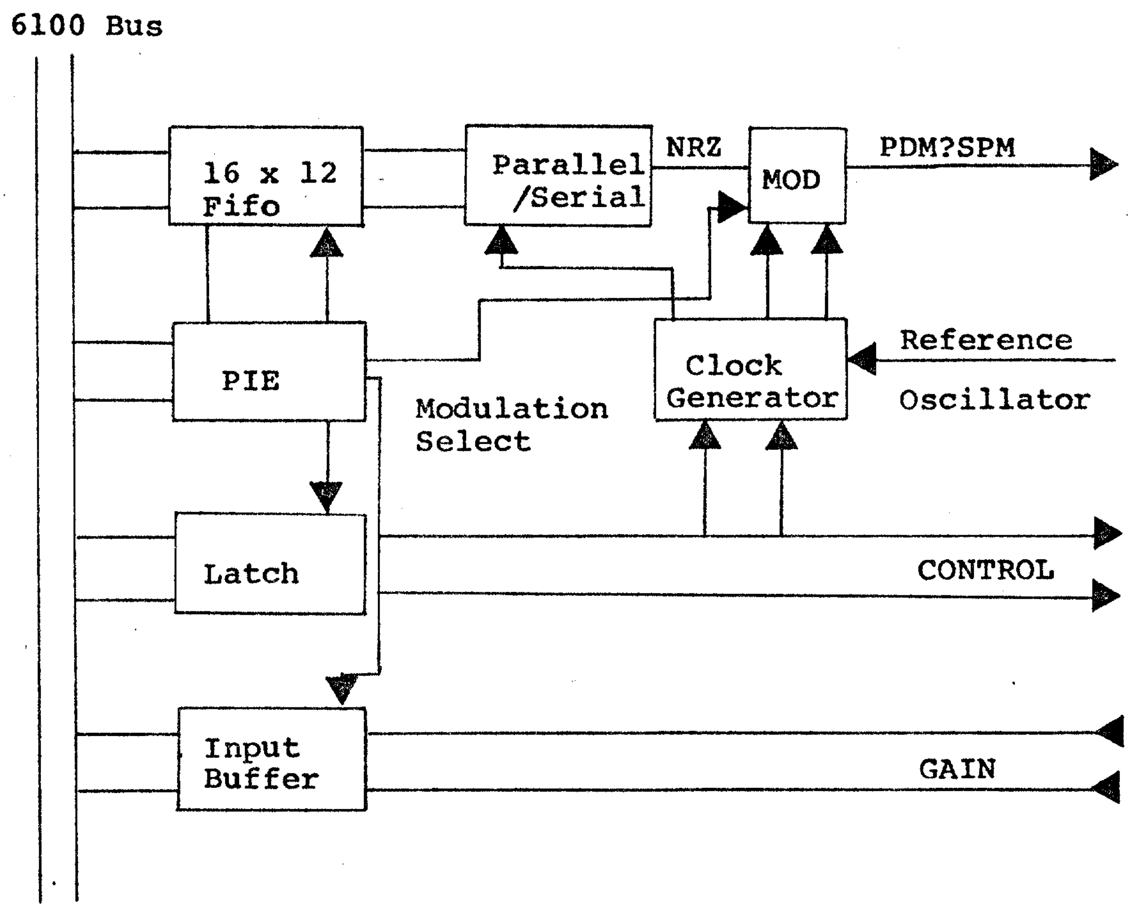


Figure 36. PWM Data Coder.

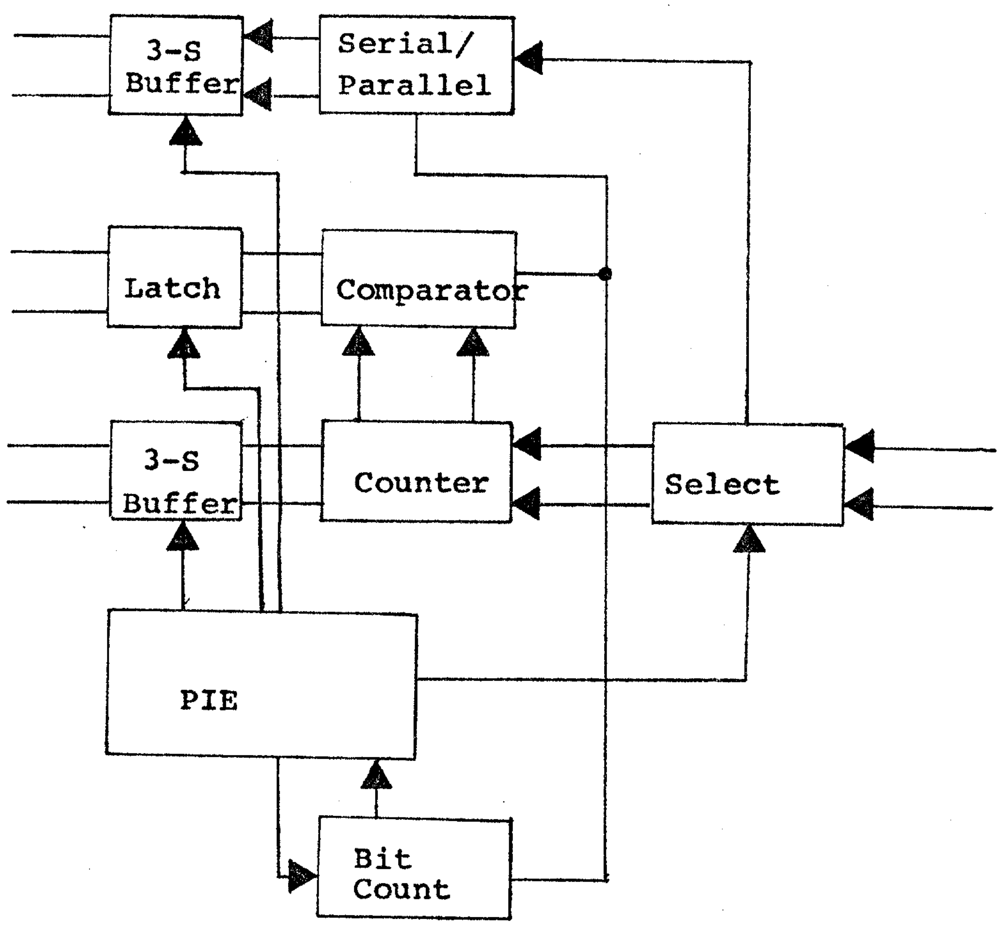


Figure 37. Data Decoder.

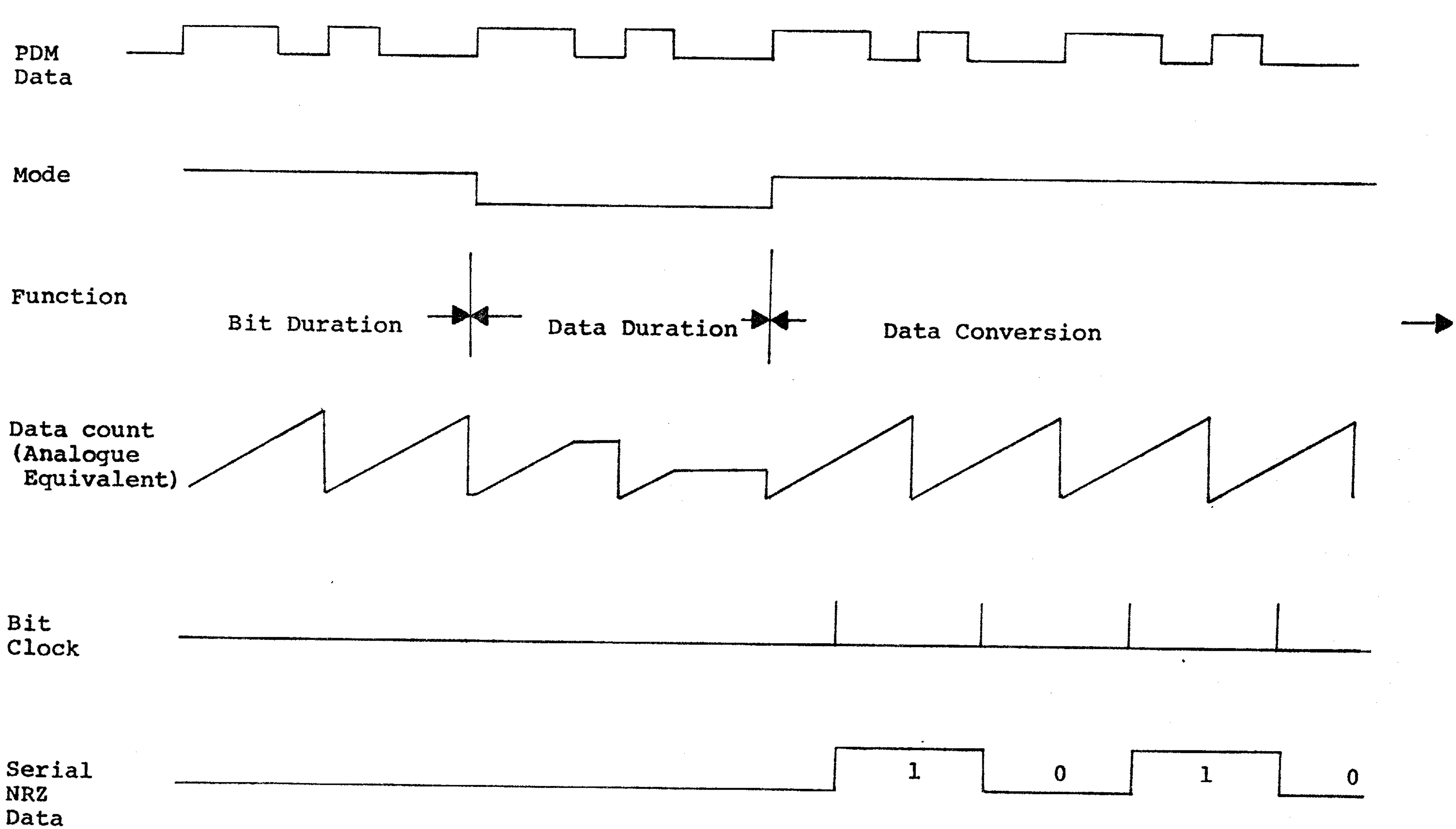


Figure 38. Data Decoder Operation.

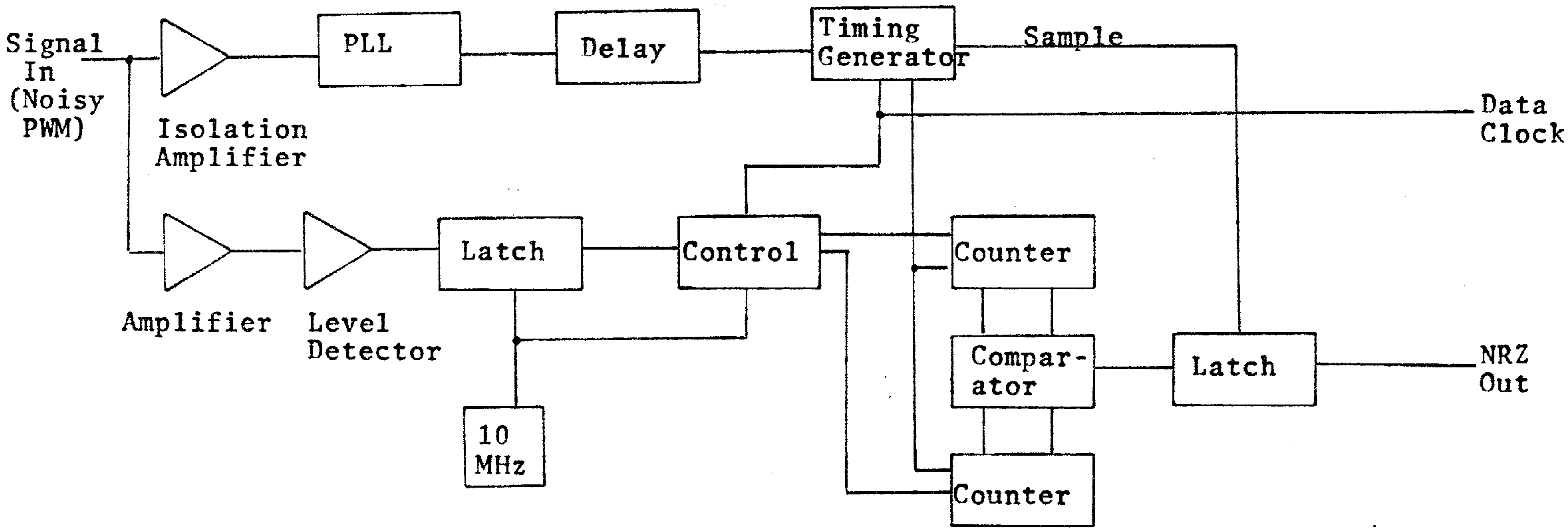


Figure 39. Probability Detector

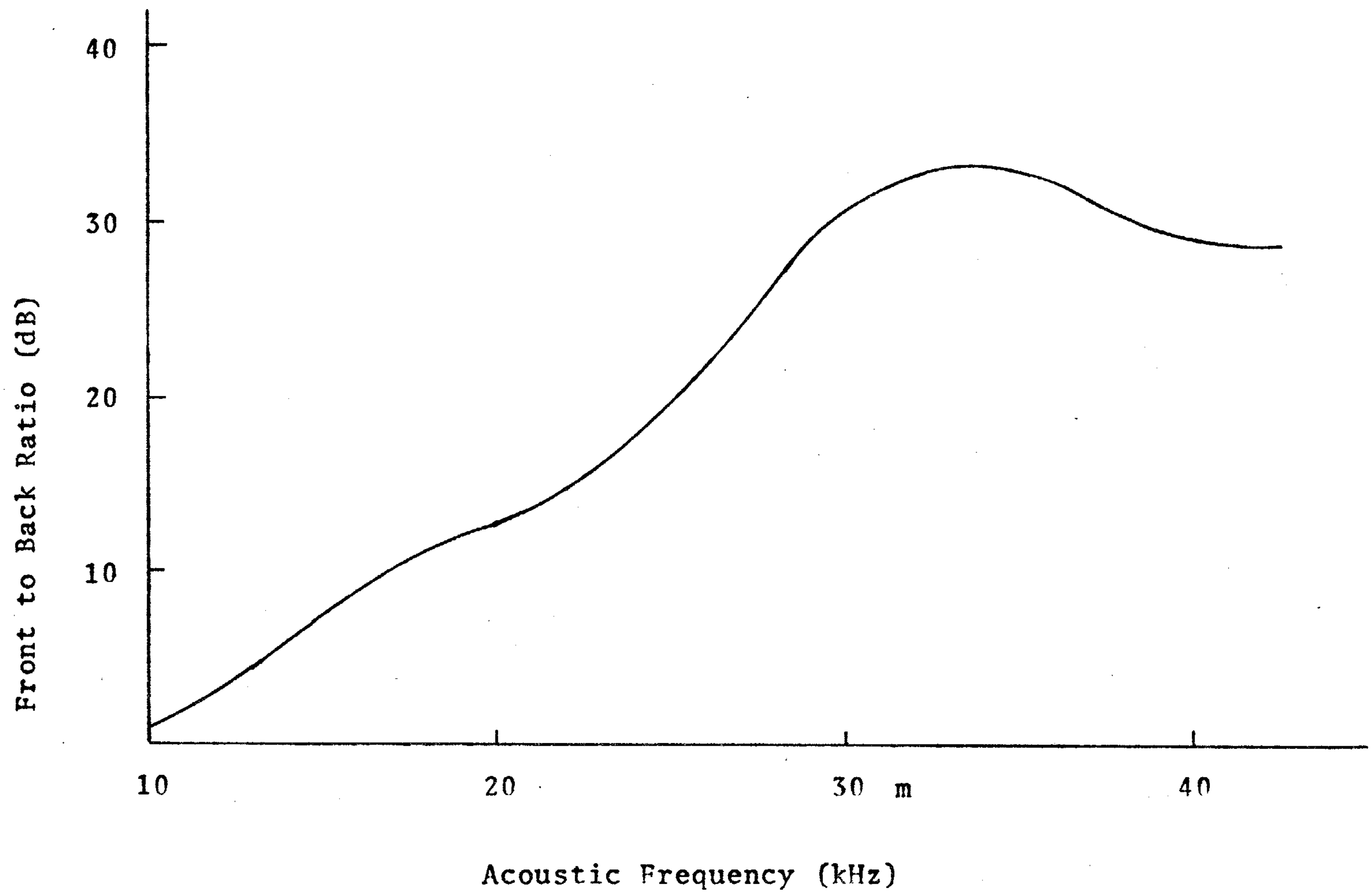


Figure 40. Example of Front to Back Ratio for Transducer/Reflector

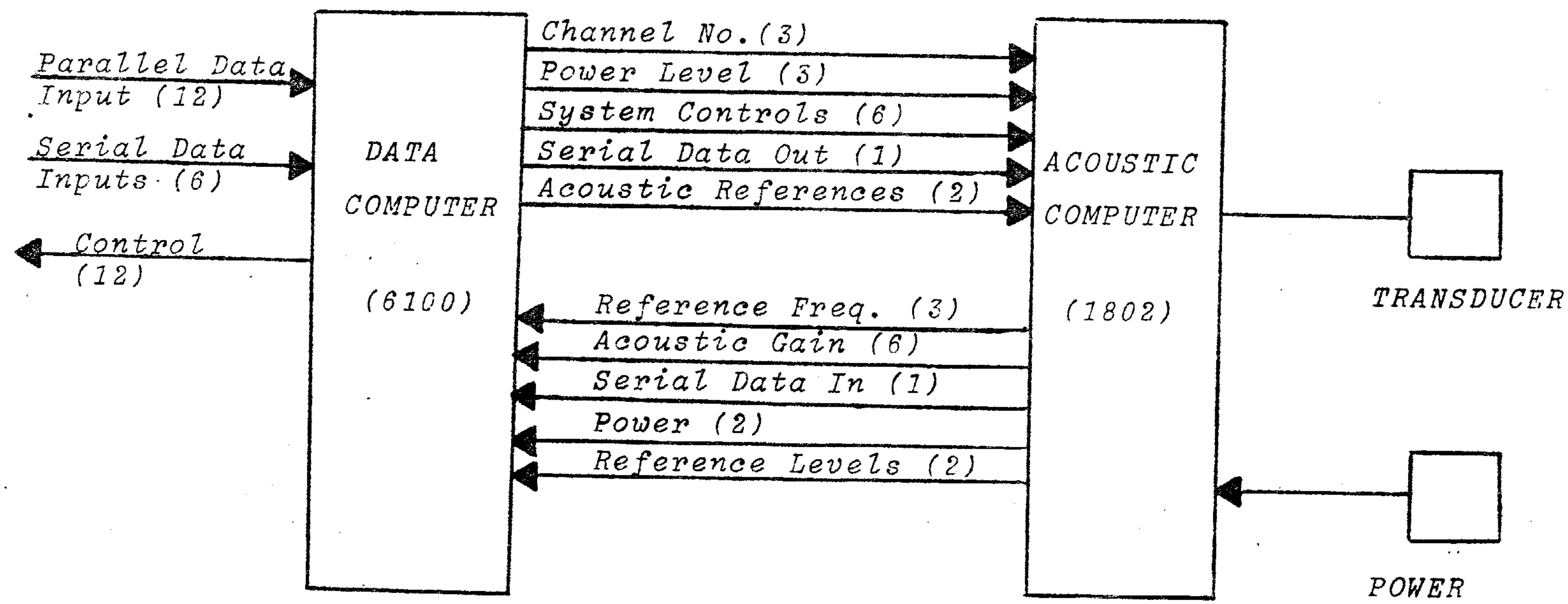


Figure 41. Transceiver Configuration.

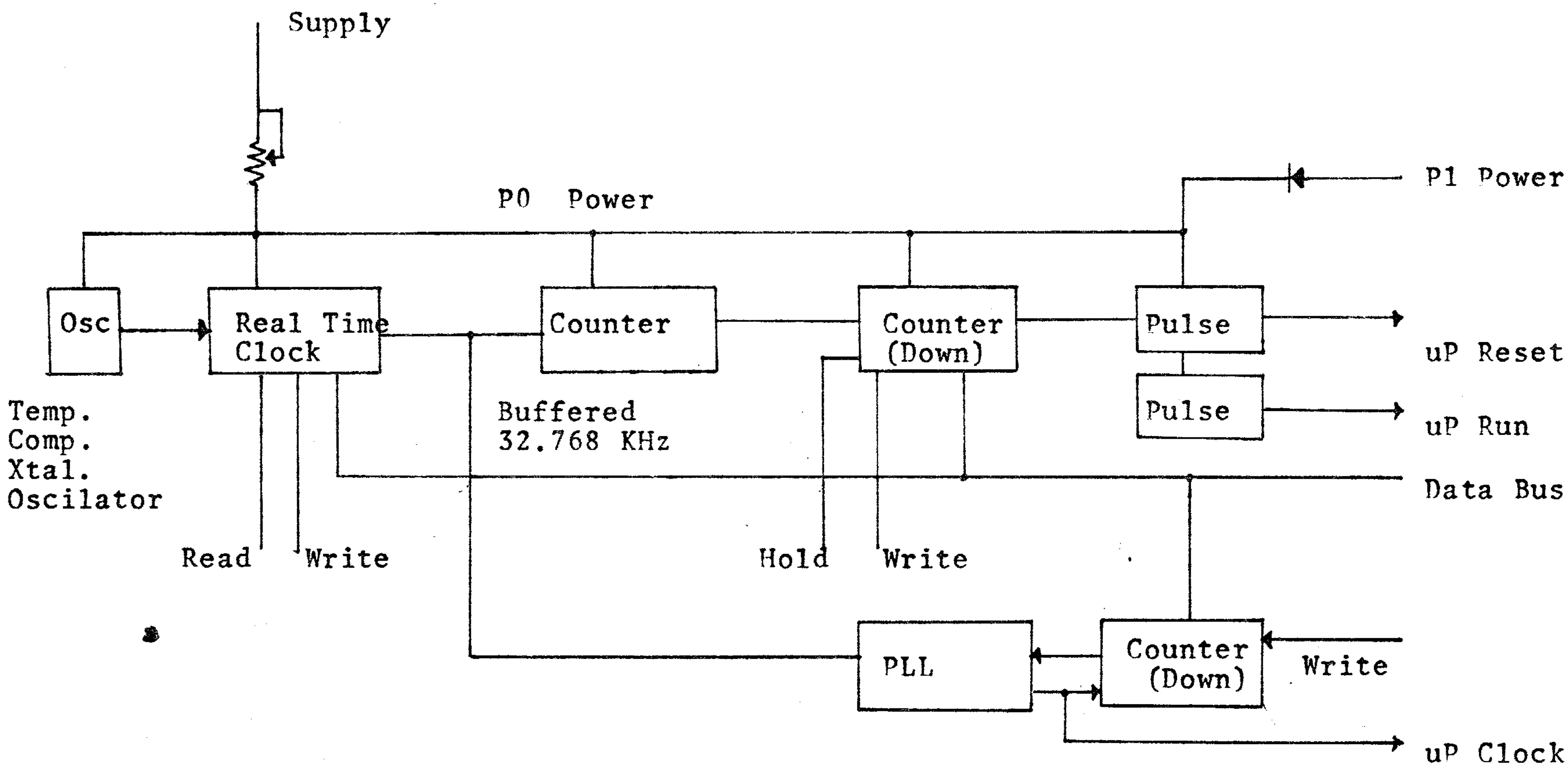


Figure 43. Clock and Calender Circuitry.

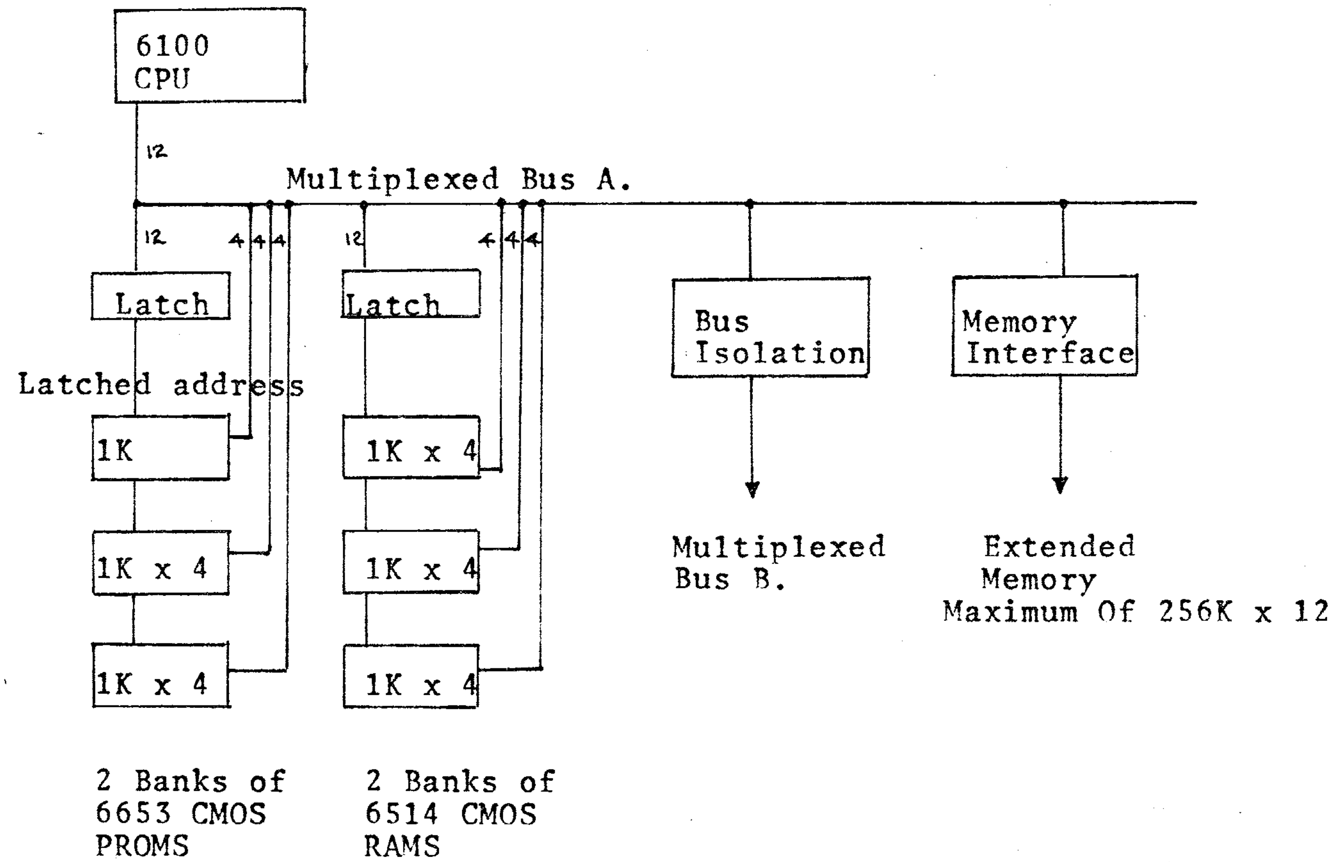


Figure 44. Arrangement For CPU and Program Memory (P1 Power)

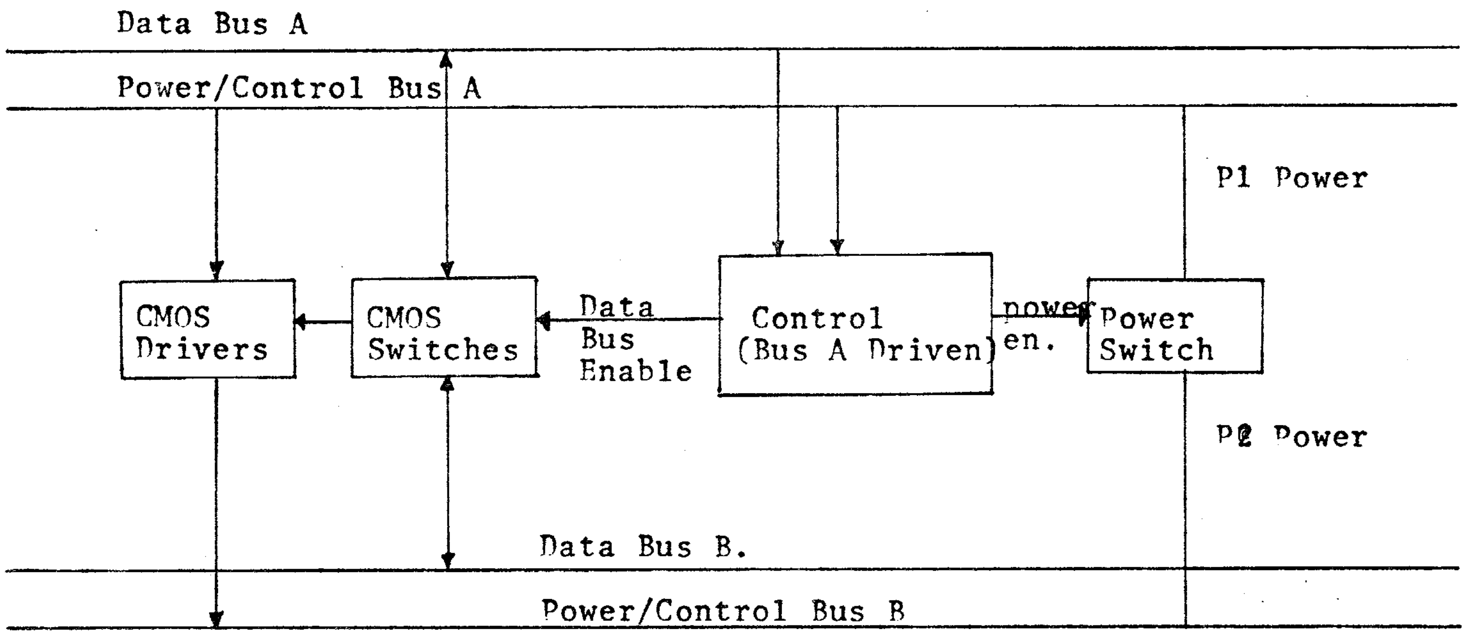


Figure 45. Bus Isolation Circuitry.

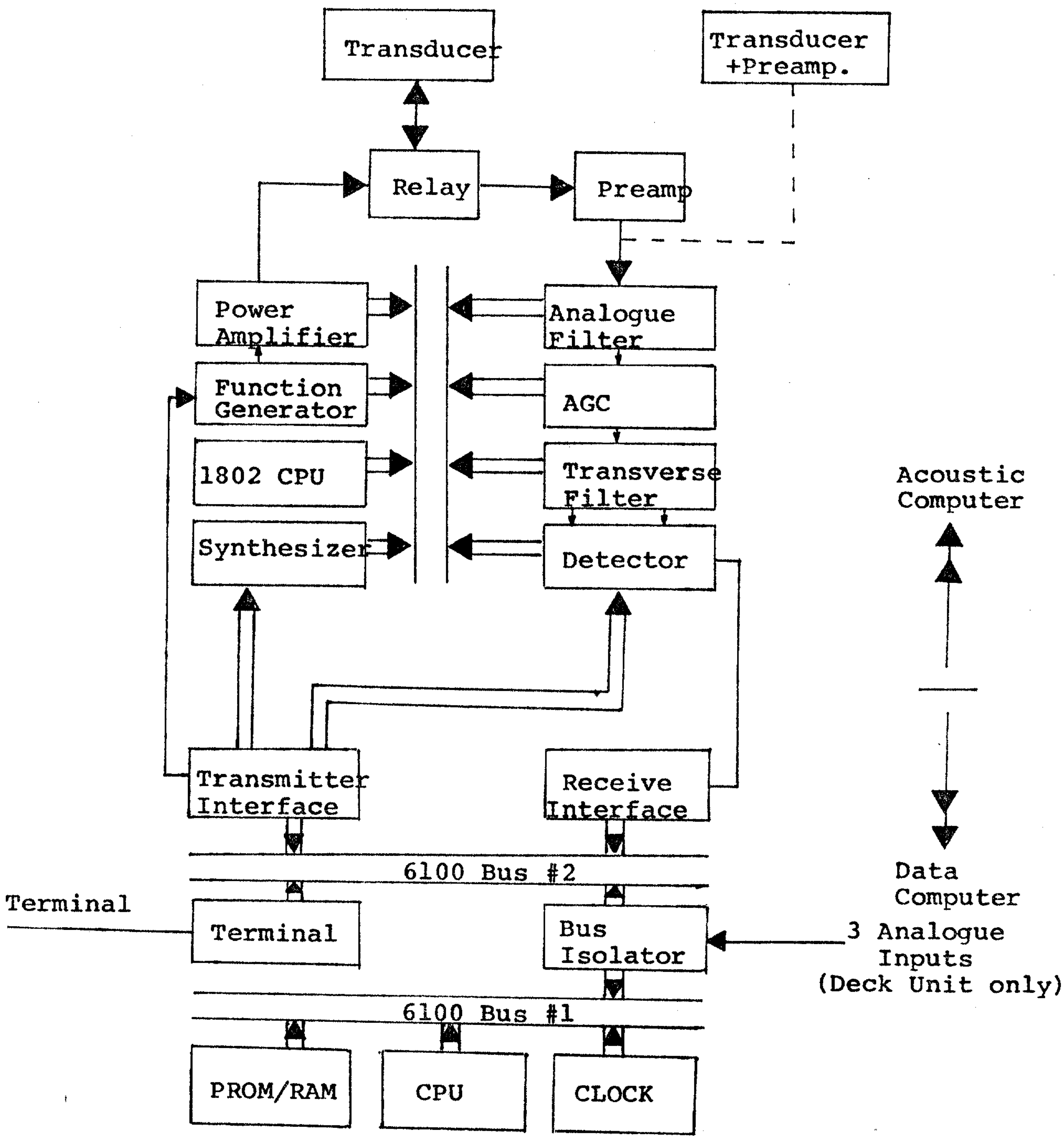


Figure 46. Acoustic Computer Configuration.

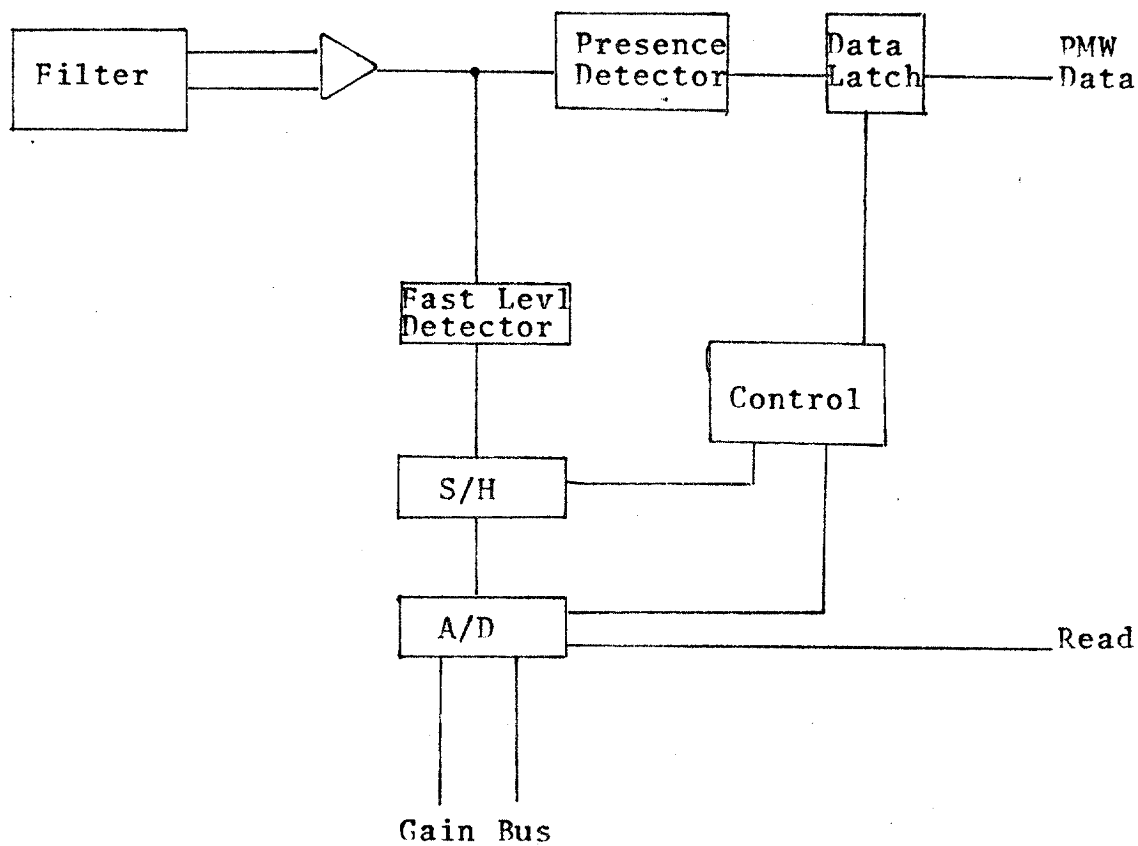


Figure 47. System Measurement of Multipath Level

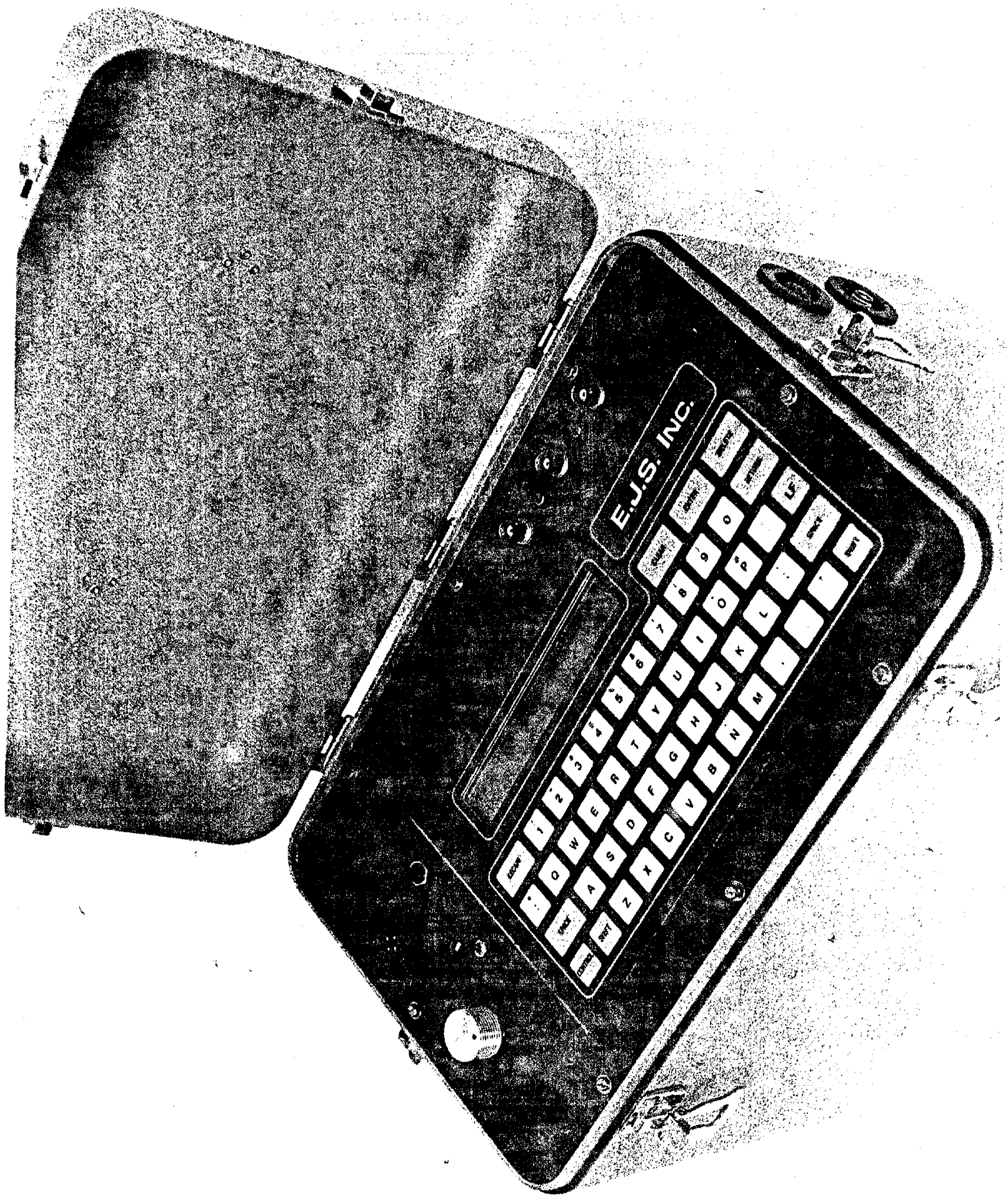


Figure 48. Typical Deck Unit Packaging.

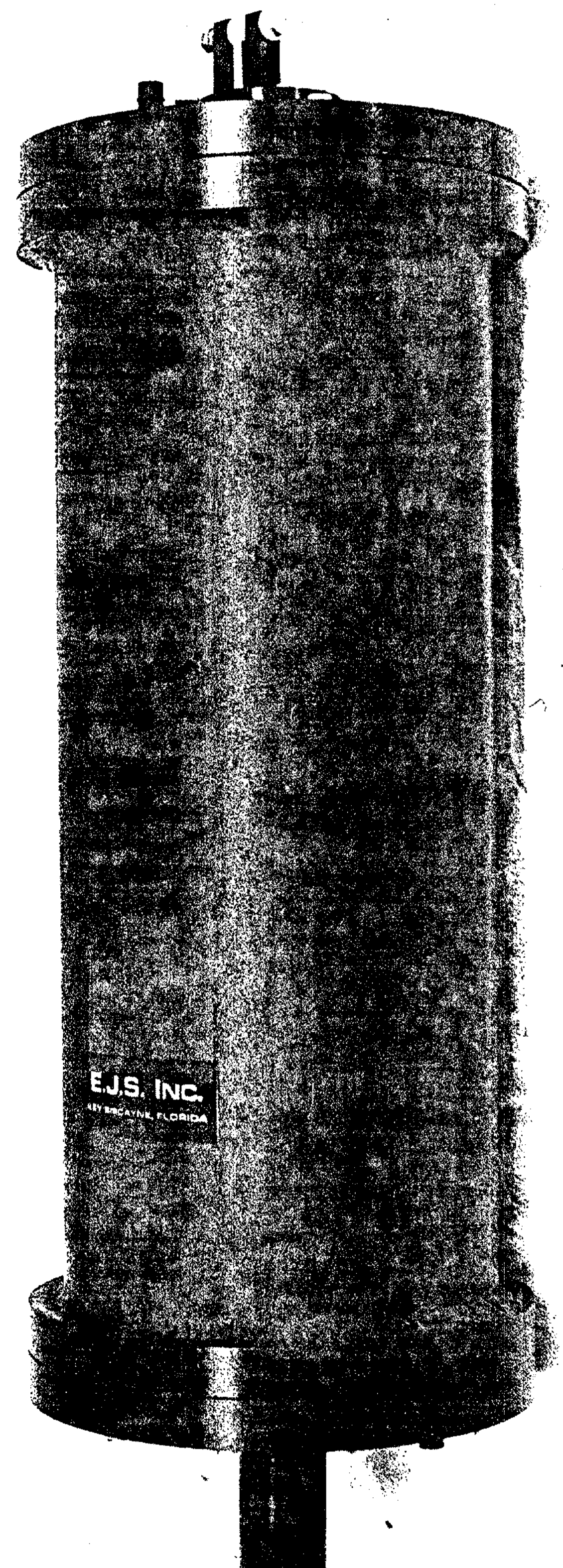
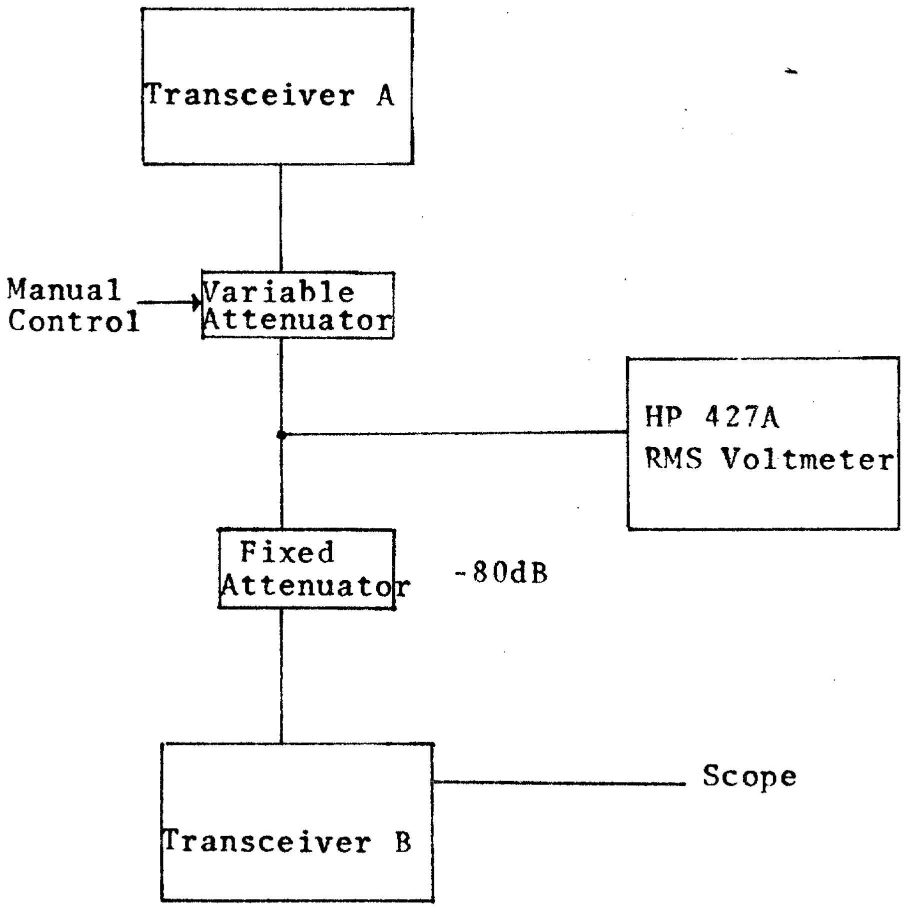


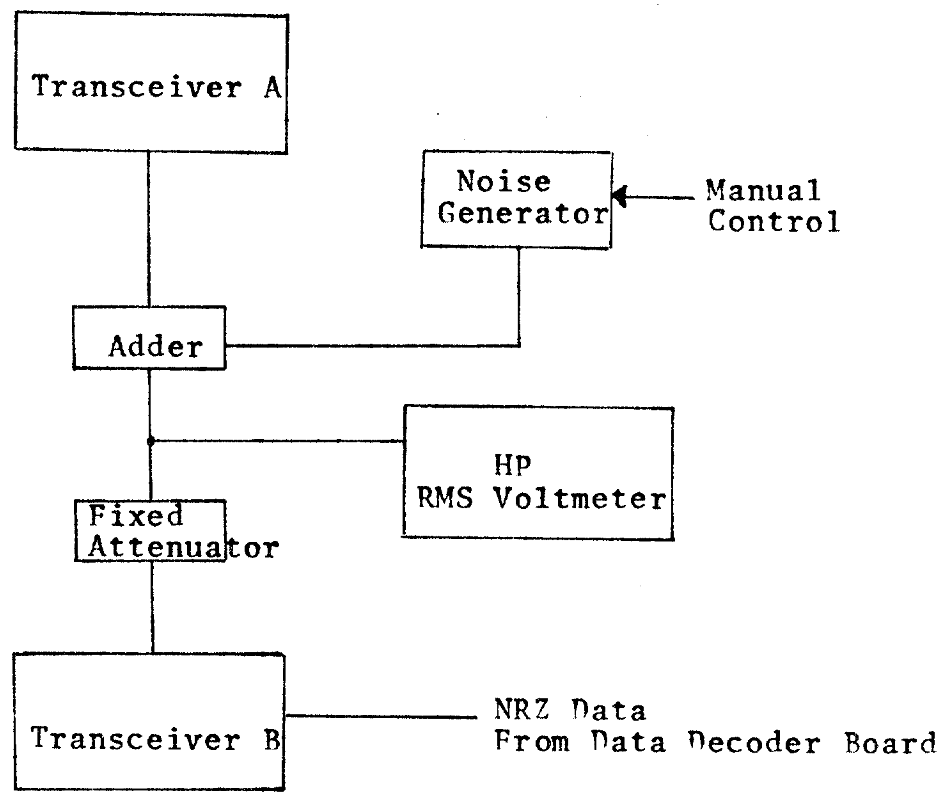
Figure 49. Typical Underwater Unit Packaging



Transceiver A - Preamble Only
Constant (low) Power

Transceiver B - Locked in Full Receive
With Data Error Analysis

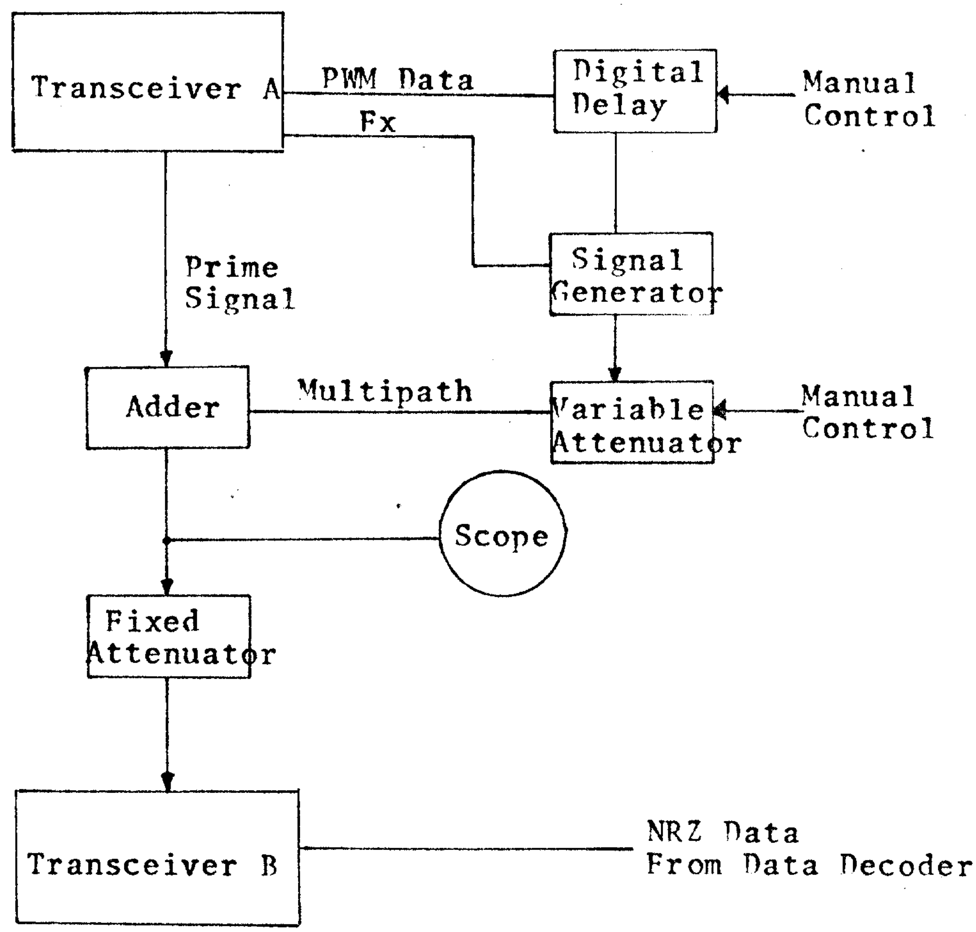
Figure 51. Receiver Sensitivity Test Setup



Transceiver A - Preamble Only
Constant (low) Power

Transceiver B - Locked in Full Receive
With Data Error Analysis

Figure 52. Receiver Signal to Noise Measurement



Transceiver A - Preamble Only
Constant (low) Power

Transceiver B - Locked in Full Receive
With Data Error Analysis

Figure 53. Receiver Signal to Multipath Test

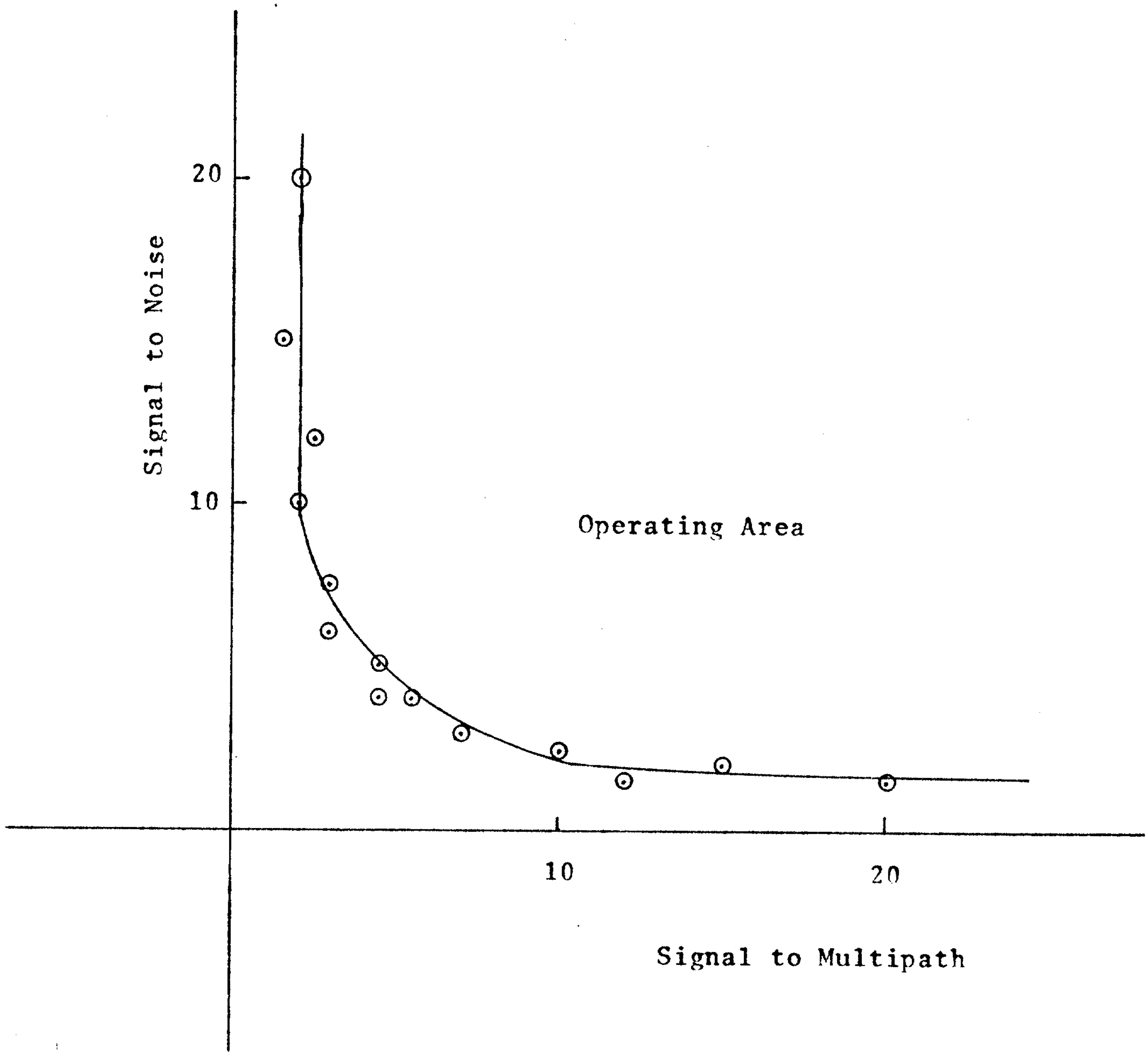


Figure 54. Effect of Combined Noise and Multipath on System Operation

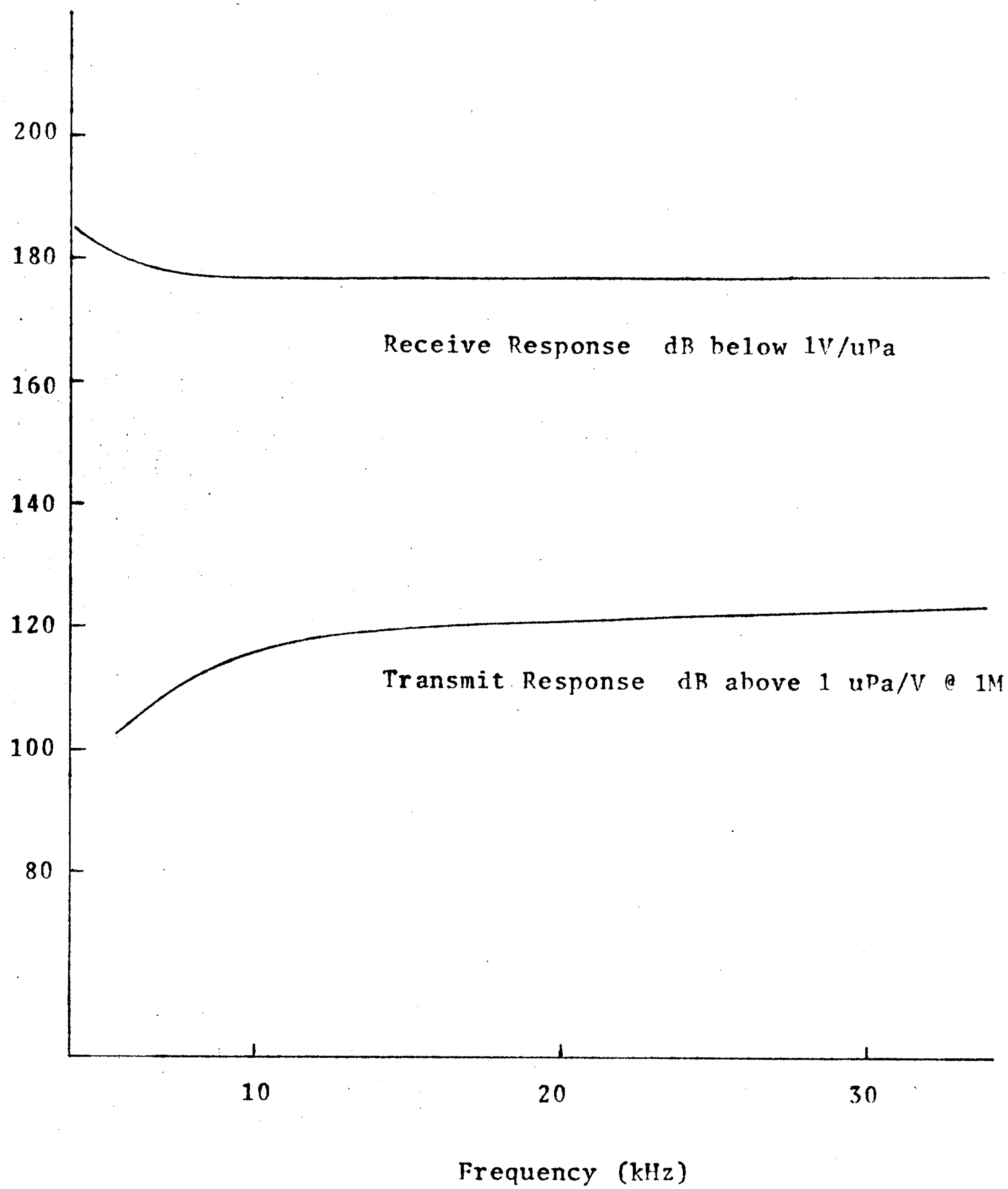


Figure 55. Cylindrical Transducer Response Functions

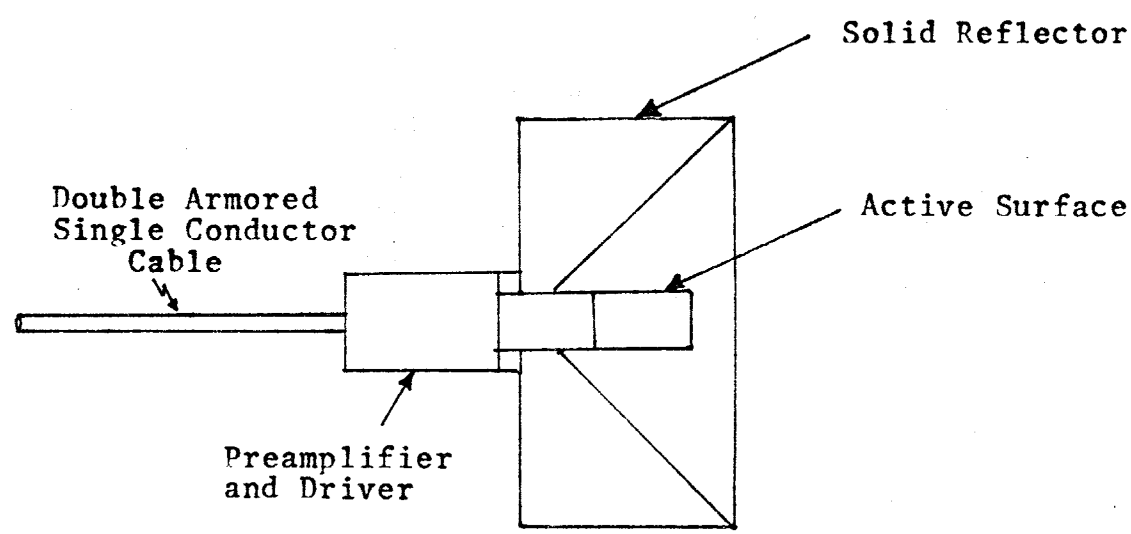


Figure 56. Transducer With Reflector

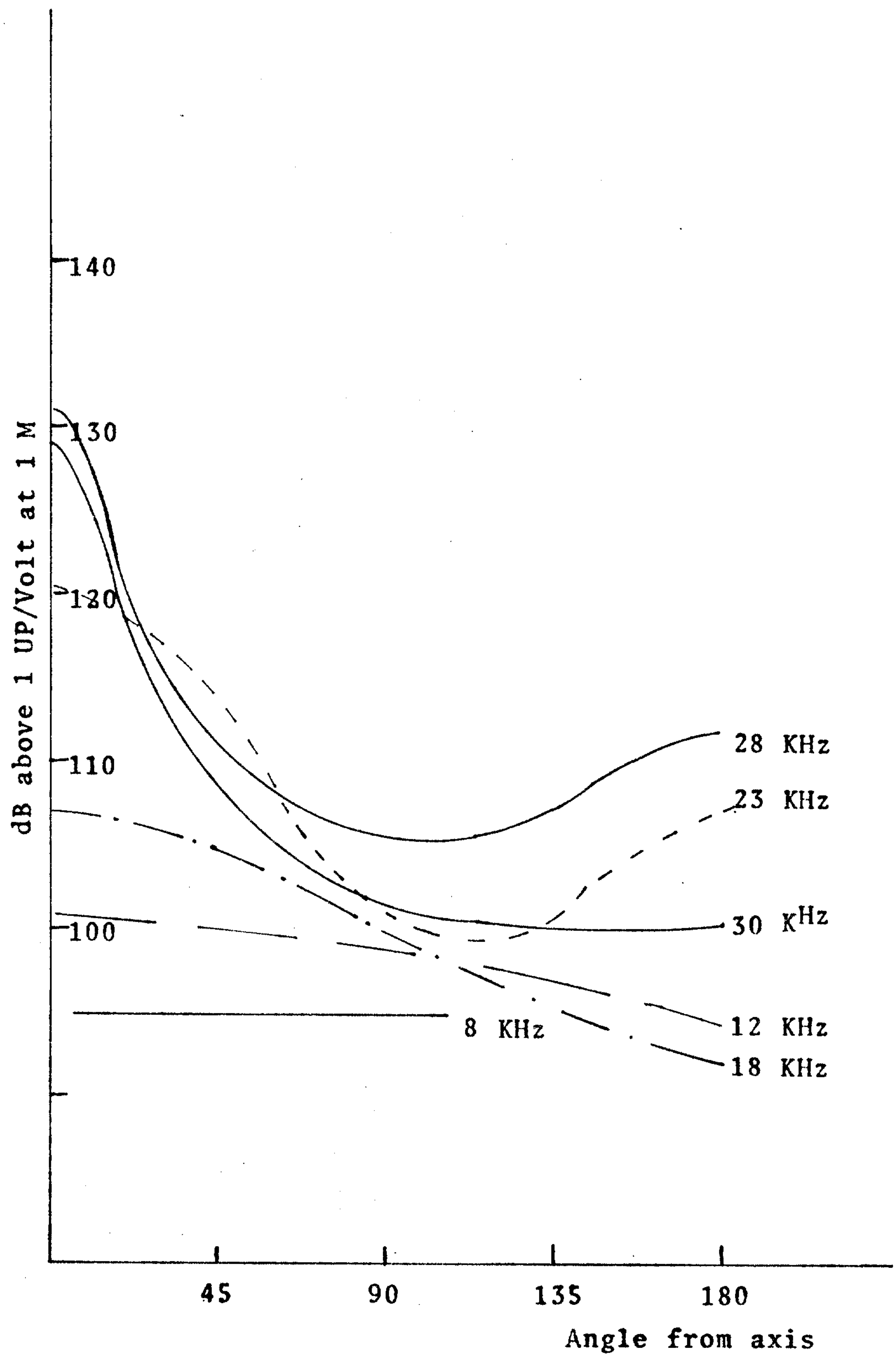
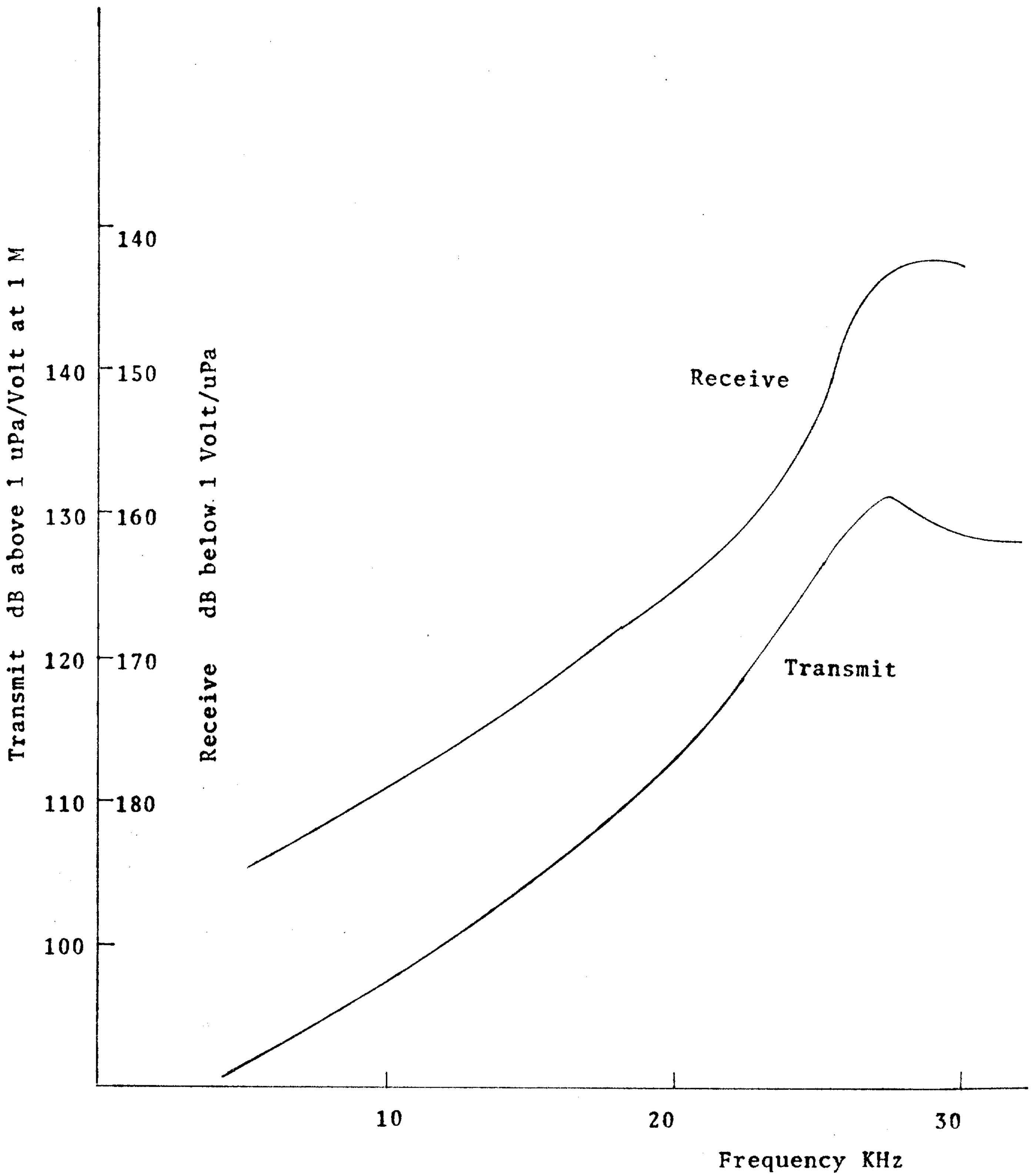


Figure 57. Directionality Of Transducer With Reflector.

Figure 58. Centerline Response Of Transducer With Reflector.



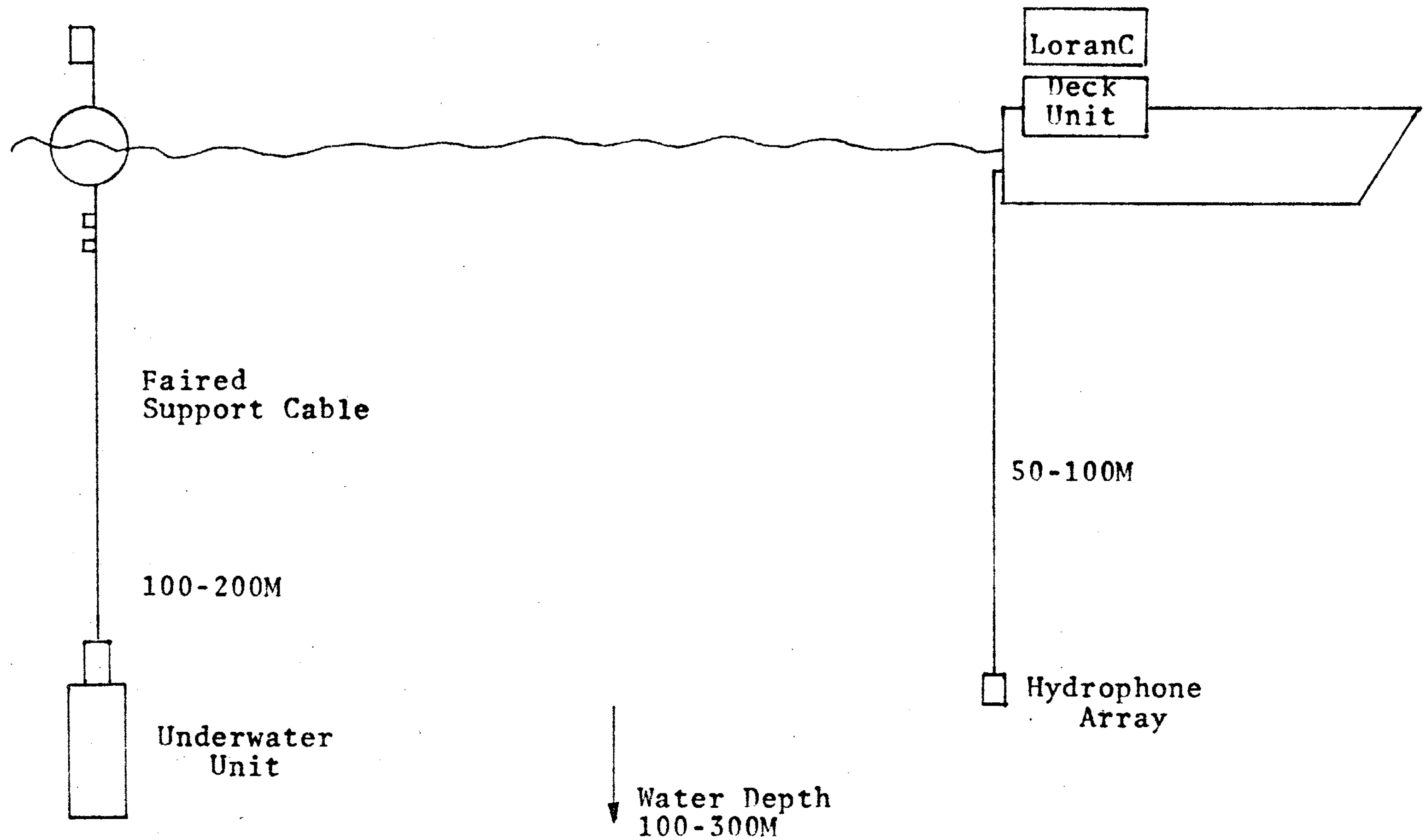


Figure 59. General Arrangement for "Float" Trials

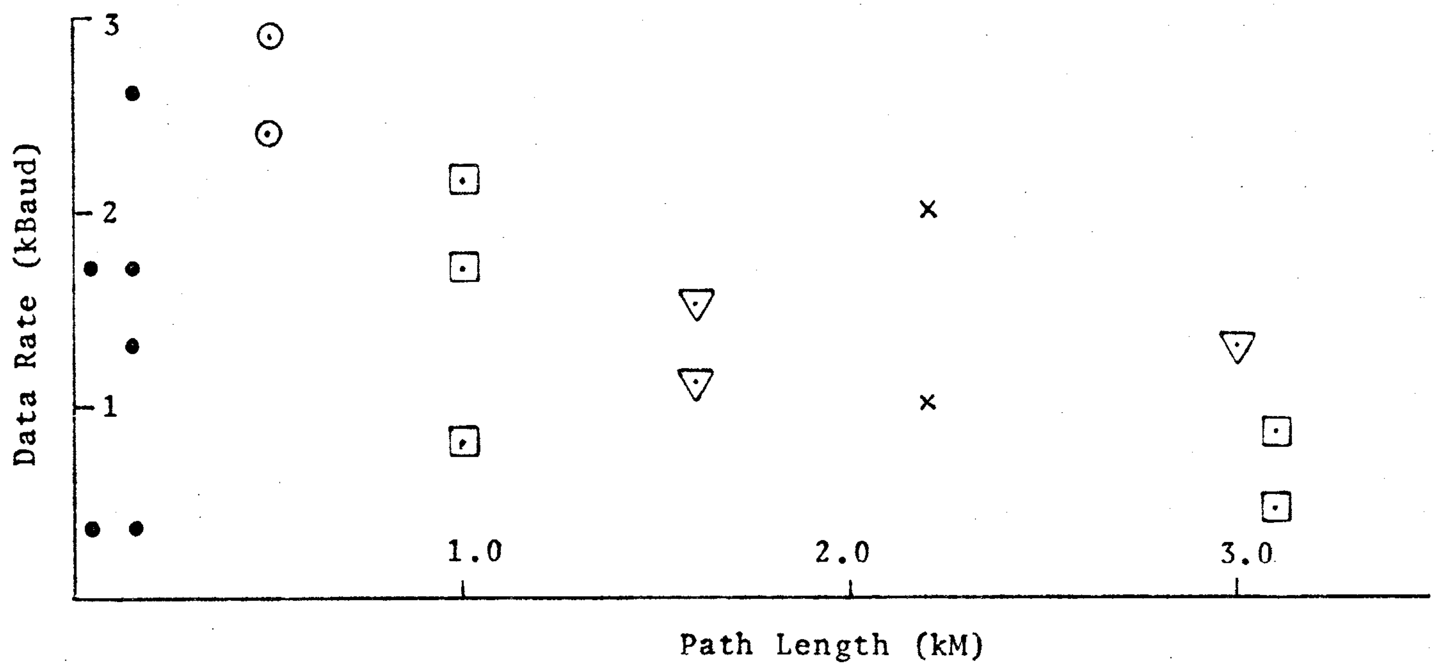


Figure 60. Summary of Data Transmissions

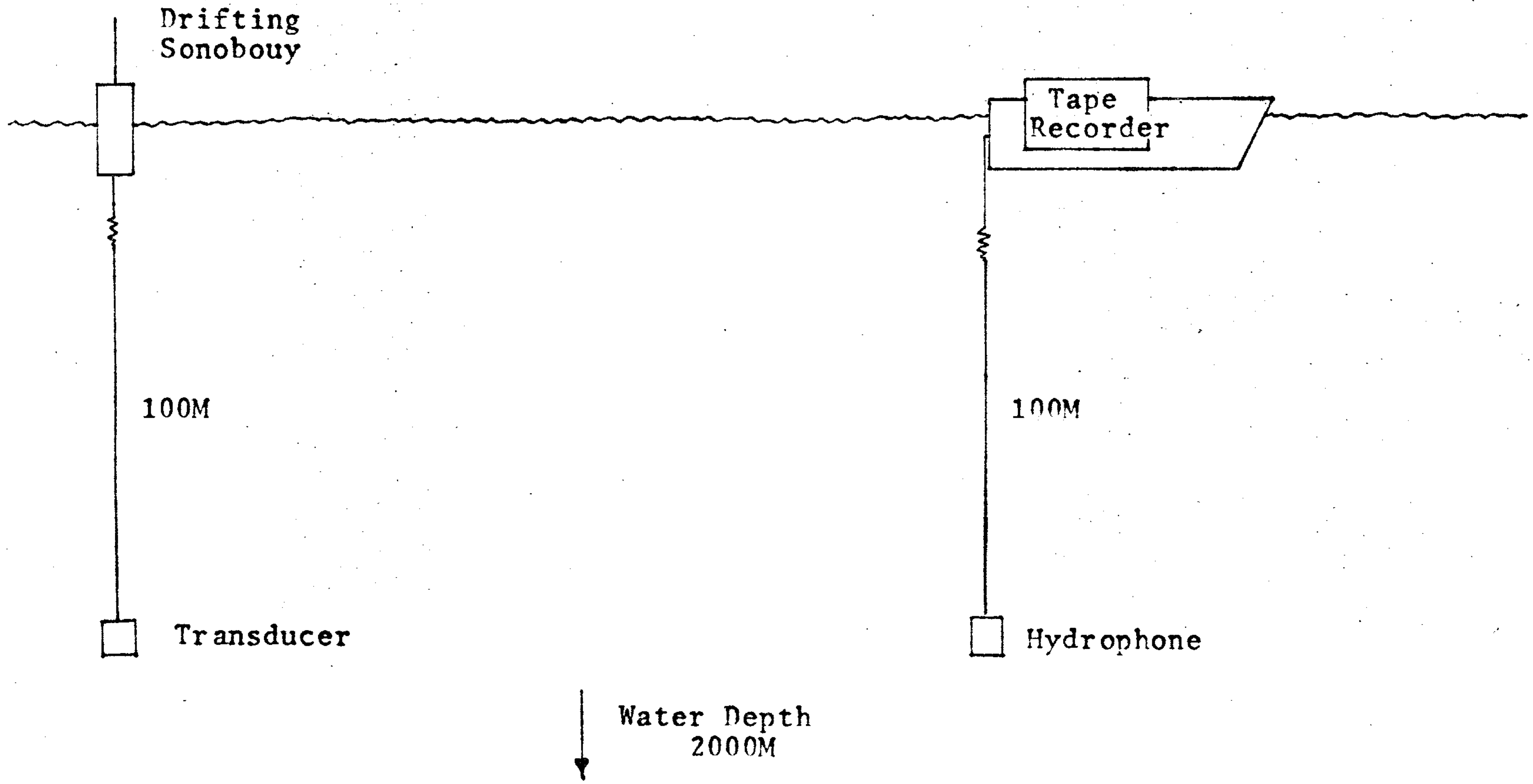


Figure 61. General Arrangement for "Sonobouy" Trial